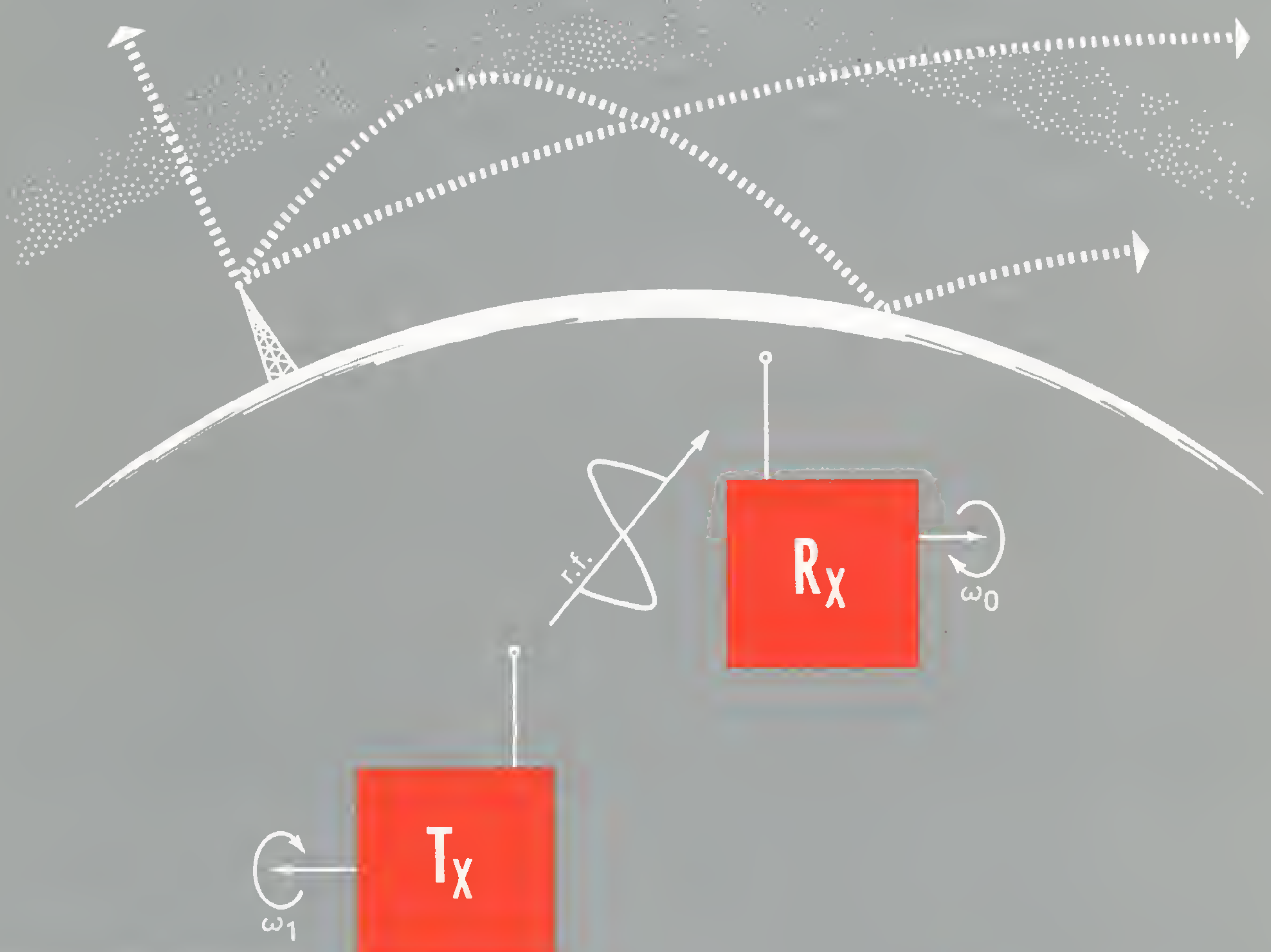
ELECTRONICS

COMMUNICATIONS



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Foreword

The marriage of electronics and technology is creating new demands for technical personnel in today's industries. New occupations have emerged with combination skill requirements well beyond the capability of many technical specialists. Increasingly, technicians who work with systems and devices of many kinds — mechanical, hydraulic, pneumatic, thermal, and optical — must be competent also in electronics. This need for combination skills is especially significant for the youngster who is preparing for a career in industrial technology.

This manual is one of a series of closely related publications designed for students who want the broadest possible introduction to technical occupations. The most effective use of these manuals is as combination textbook-laboratory guides for a full-time, post-secondary school study program that provides parallel and concurrent courses in electronics, mechanics, physics, mathematics, technical writing, and electromechanical applications.

A unique feature of the manuals in this series is the close correlation of technical laboratory study with mathematics and physics concepts. Each topic is studied by use of practical examples using modern industrial applications. The reinforcement obtained from multiple applications of the concepts has been shown to be extremely effective, especially for students with widely diverse educational backgrounds. Experience has shown that typical junior college or technical school students can make satisfactory progress in a well-coordinated program using these manuals as the primary instructional material.

School administrators will be interested in the potential of these manuals to support a common first-year core of studies for two-year programs in such fields as: instrumentation, automation, mechanical design, or quality assurance. This form of *technical core* program has the advantage of reducing instructional costs without the corresponding decrease in holding power so frequently found in general core programs.

This manual, along with the others in the series, is the result of six years of research and development by the *Technical Education Research Centers, Inc.*, (TERC), a national nonprofit, public service corporation with headquarters in Cambridge, Massachusetts. It has undergone a number of revisions as a direct result of experience gained with students in technical schools and community colleges throughout the country.

Maurice W. Roney

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Preface

When we think of electronic communications it is natural to think first of television because we see it so frequently. While television is an extremely important part of the communication area, we should not limit our thinking to it alone.

If we take electronic communications to mean the conveying of information from point-to-point by means of electromagnetic waves, then we must consider many applications other than television. Radiotelephony and radiotelegraphy are perhaps the most obvious of these other applications. Less obvious, but nevertheless extremely important, are the various types of radar, telemetry, and remote control applications.

Each of these many application areas is a technical specialty within itself. We could never hope to cover all of them in detail within a single volume.

There are, however, a number of basic communication principles which are shared by all of these application areas. In this volume we shall cover a number of these basic areas including: receivers, transmitters, and introductory communications systems.

Try to keep in mind that while a principle may be illustrated by a specific application, the principle applies to many other applications as well.

The sequence of presentation chosen is by no means inflexible. It is expected that individual instructors may choose to use the materials in other than the given sequence.

The particular topics chosen for inclusion in this volume were selected primarily for convenience and economy of materials. Some instructors may wish to omit some of the exercises or to supplement some of them to better suit their local needs.

The materials are presented in an action-oriented format combining many of the features normally found in a textbook with those usually associated with a laboratory manual. Each experiment contains:

1. An INTRODUCTION which identifies the topic to be examined and often includes a rationale for doing the exercise.

2. A DISCUSSION which presents the background, theory, or techniques needed to carry out the exercise.

3. A MATERIALS list which identifies all of the items needed in the laboratory experiment. (Items usually supplied by the student such as pencil and paper are not normally included in the lists.)

4. A PROCEDURE which presents step-by-step instructions for performing the experiment. In most instances the measurements are done before calculations so that all of the students can at least finish making the measurements before the laboratory period ends.

5. An ANALYSIS GUIDE which offers suggestions as to how the student might approach interpretation of the data in order to draw conclusions from it.

6. PROBLEMS are included for the purpose of reviewing and reinforcing the points covered in the exercise. The problems may be of the numerical solution type or simply questions about the exercise.

Students should be encouraged to study the textual material, perform the experiment, work the review problems, and submit a technical report on each topic. Following this pattern, the student can acquire an understanding of, and skill with, basic communications circuits that will be extremely valuable on the job.

These materials on basic communications comprise one of a series of sixteen volumes prepared for technical students by the TERC EMT staff at Oklahoma State University, under the direction of Dr. D.S. Phillips and R. W. Tinnell. The principal authors of these materials were Terry E. Schultz and R. W. Tinnell.

An *Instructor's Data Guide* is available for use with this volume. Mr. Kenneth F. Cathey was responsible for testing the materials and compiling the instructor's data guide for them. Other members of the TERC staff made valuable contributions in the form of criticisms, corrections, and suggestions.

It is sincerely hoped that this volume as well as the other volumes in this series, the instructor's data books, and the other supplementary materials will make the study of technology interesting and rewarding for both students and teachers.

Contents

experiment 1 *RADIO RECEIVERS*

INTRODUCTION. Communications is perhaps the cornerstone of all electronics. A communications device that is very familiar to most people is the receiver. In this experiment we shall examine some common receiver systems.

DISCUSSION. Electronic communications is usually accomplished by means of radio frequency links. The information which is communicated may be of any type. For instance, it could be sound (as in the case of a commercial radio station), moving pictures (television), still pictures (facsimiles), positional data (radar), or any one of several other types of information. For the purposes of simplicity, we shall identify any form of radio frequency communications with the general term radio. Remember that we are not limiting the term radio to sound communications alone.

In most cases the information to be communicated is of a relatively low frequency and will not radiate effectively from an antenna. To achieve effective radiation we normally superimpose the information to be communicated upon a relatively high-frequency RF carrier.

Consequently, a radio signal is composed of two parts. They are:

1. The information to be communicated called the modulation signal or more simple the modulation.
2. The high-frequency RF carrier signal.

Later we will examine the business of combining the carrier and modulation signals. For the time being, let's just note that there are two main types of radio signals. One of the types is an *AM radio* signal. The AM stands for *amplitude modulation.* In an AM system the modulation signal is combined with the carrier in such a way that the amplitude of the carrier varies with the modulation. Figure 1-1 shows a schematic representation of an AM signal.

The other principal type of modulation is *FM* or *frequency modulation.* In this type of modulation the two signals are combined in such a way that the frequency of the carrier varies with the amplitude of the modulation. Figure 1-2 shows this in pictorial form.

Due to the nature of the radio signal, a receiver must do four operations. These basic receiver operations are:

(a) Reception of radio signals.
(b) Selection of the single desired signal.
(c) Separation of the modulation from the carrier.
(d) Reproduction of the modulation.

Fig. 1-1 Amplitude Modulation

Fig. 1-2 Frequency Modulation

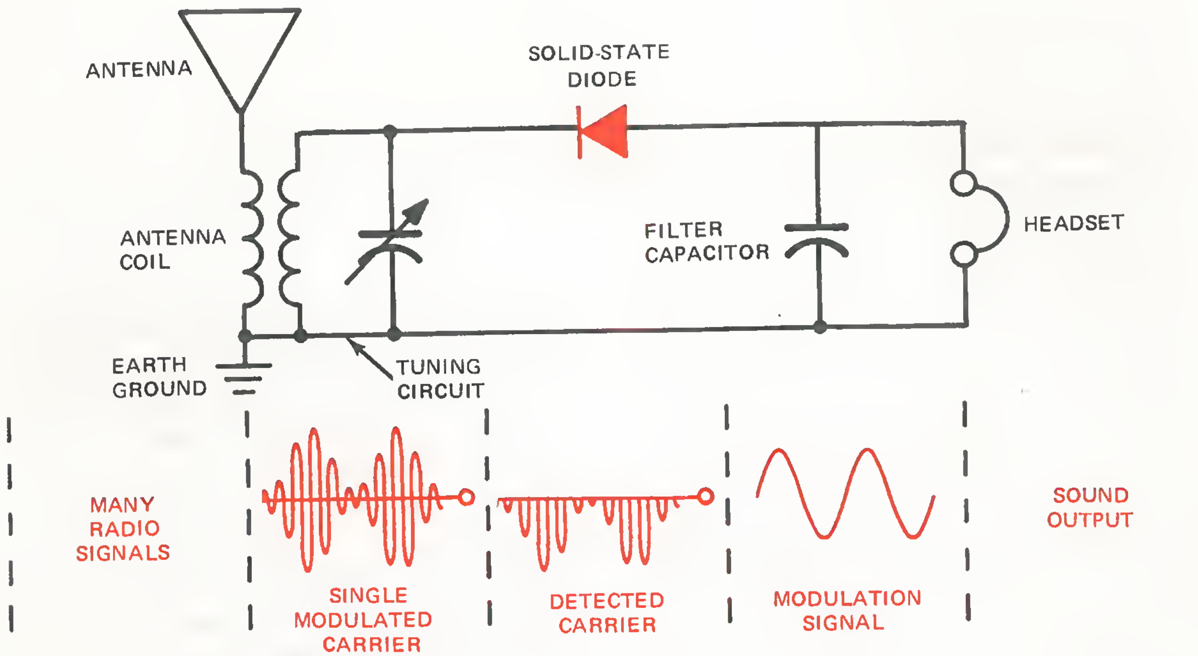


Fig. 1-3 A Simple Radio Receiver (AM)

Probably the simplest radio receiver circuit that will perform these jobs is the "crystal" radio shown in figure 1-3.

The antenna and antenna coil together with the ground connection perform the reception function. Many different radio signals may cause some current to flow in the antenna coil. The tuning circuit selects one of these signals and conveys it to the diode. The diode detects or rectifies the signal. The filter capacitor provides a low impedance path to ground for the high-frequency carrier signal. The low-frequency modulation signal passes through the headset (earphone) and can be heard.

In the early days of radio, many people built radios very much like this one and used them. They used a galena crystal and often wound their own coils.

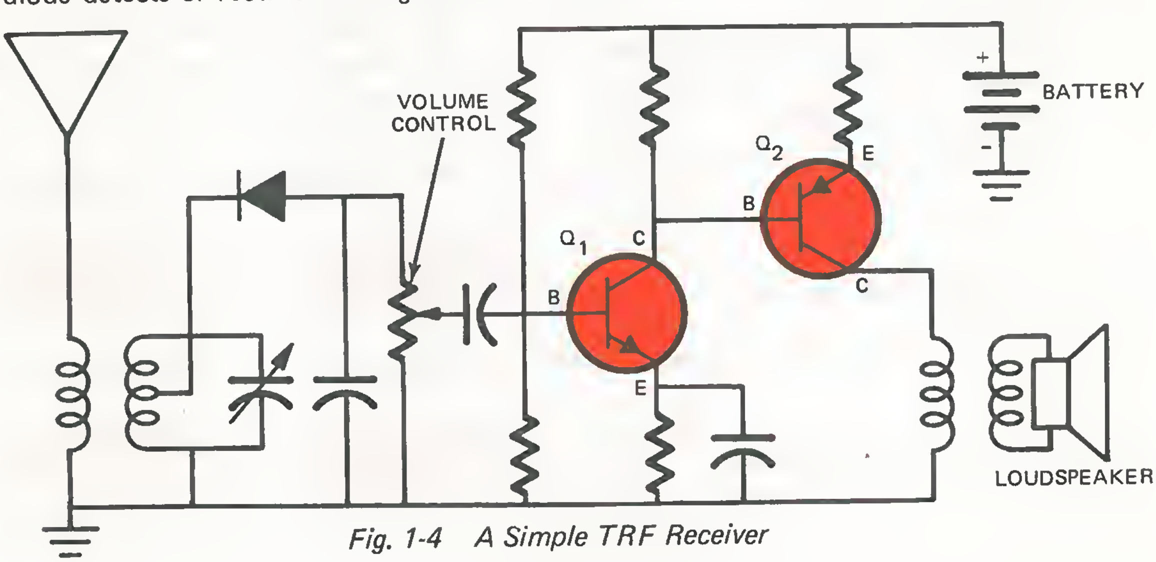


Fig. 1-4 A Simple TRF Receiver

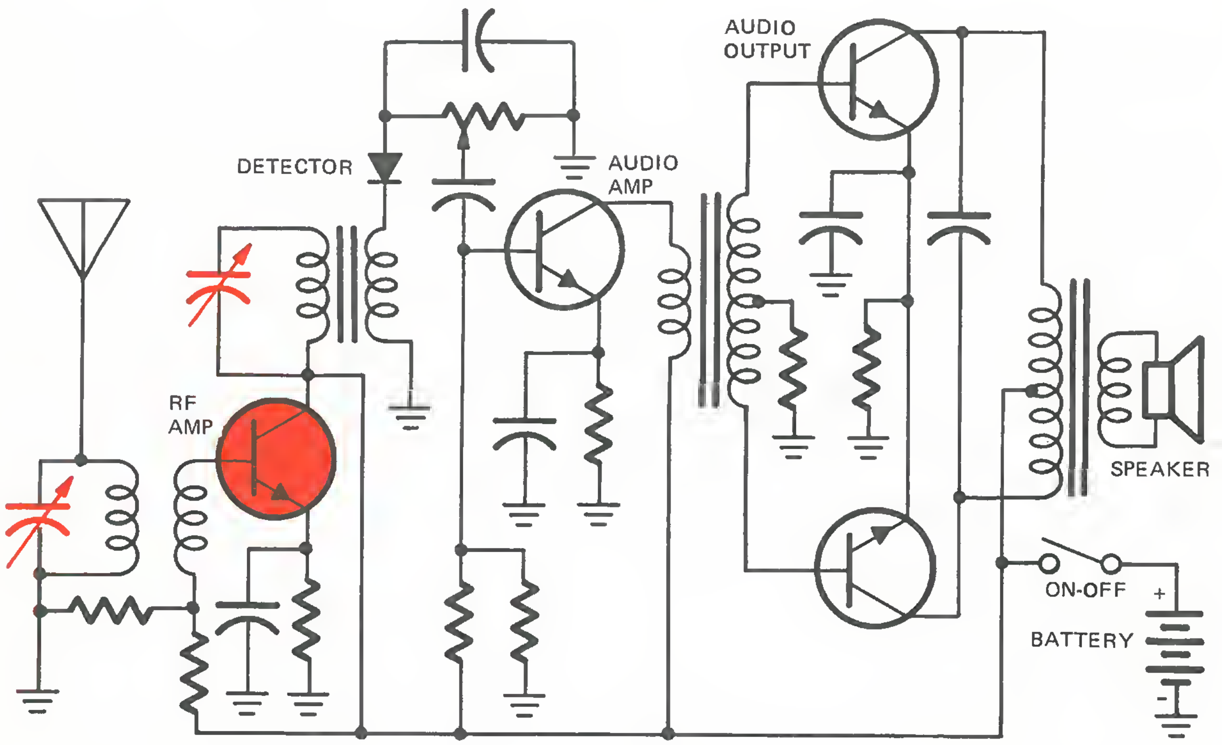


Fig. 1-5 A TRF Receiver

It wasn't long, however, before people wanted more powerful outputs. To get more output we can simply add an audio amplifier section as shown in figure 1-4. The operation is still the same but now we have a loudspeaker and a volume control. Some very low cost consumer receivers are still built using circuitry similar to this one. This type of receiver is called a *tuned radio frequency* receiver or more simply a *TRF.*

Using a tapped coil instead of one like that used in figure 1-3 helps to match the detector impedance to that of the tuning circuit. As a result, greater *sensitivity* (the ability to receive small signals) is achieved.

Also, since there is less loading on the coil, the tuning circuit has a higher effective Q. This high Q allows the tuning circuit to tune more narrowly, thereby providing greater *selectivity* (the ability to select one of several signals that are near the same frequency).

To achieve even greater sensitivity and selectivity, one might decide to use a stage or two of RF amplification before the detector. Figure 1-5 shows a receiver with RF amplification.

This idea was, in fact, widely used in the early days. You have, no doubt, seen some of the old time radios that had five or six tuning knobs on the front. In those days each stage was tuned separately and many of them had four or five RF stages. It took considerable skill and patience to tune in a station correctly.

Later, tuning capacitors were built which had several sections "ganged" on the same shaft. These multiple-gang capacitors allowed receivers to be built with only one tuning knob.

The receiver shown in figure 1-5 would need a two-gang capacitor — one for the antenna tuning and one for the RF amplifier.

TRF circuits are still used in some of the more economical consumer radios.

Another method of increasing sensitivity and selectivity is to use positive feedback in the RF amplifier circuit. A receiver which uses this feature is called a *regenerative receiver.* Figure 1-6 shows a simple regenerative circuit. Feedback is achieved by means of a *tickler coil* in the RF amplifier collector circuit. Regeneration substantially increases the gain of the RF amplifier but is somewhat susceptible to the uncontrolled oscillation.

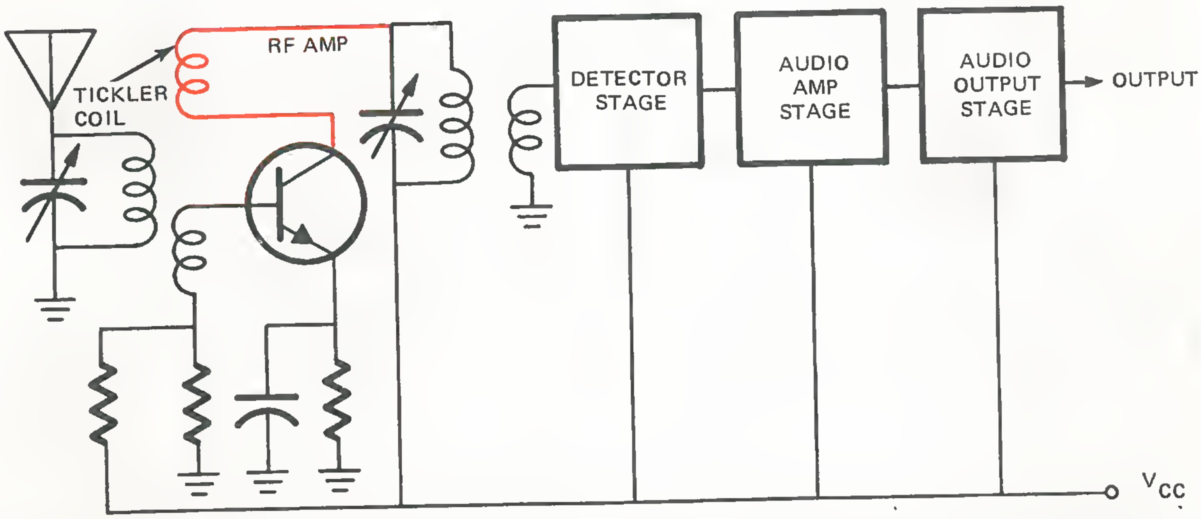


Fig. 1-6 A Regenerative Receiver

Yet another method of increasing sensitivity and selectivity is to feed back both the RF and the audio signal. The circuit in figure 1-7 shows this type of receiver circuit. In this circuit the RF signal is selected and amplified by Q_1. The antenna

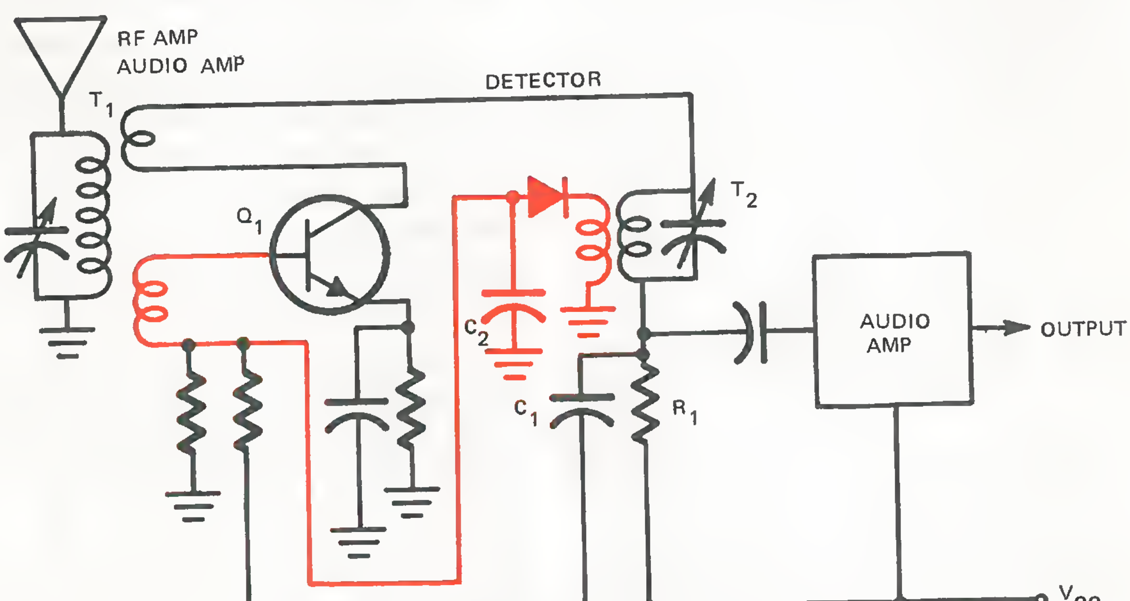


Fig. 1-7 A Regenerative-Reflex Receiver

transformer T_1 provides RF regeneration via the tickler coil. The RF collector load T_2 couples the signal to the detector. C_2 is the detector filter. After detection the modulation is fed back to the base of Q_1. The transistor now amplifies the modulation signal. In the collector circuit the RF load components T_1 and T_2 are so small in value that the modulation signal only sees R_1 as a load. The RF signal does not see R_1 because of the bypass capacitor C_1.

Finally, the modulation signal is coupled out to the audio amplifier for further amplification.

Receivers of this type, in which one stage handles both the RF and the modulation signals, are called *reflex* circuits. They are sometimes used for low cost consumer radios and special purpose receivers.

Of these receivers that we have considered so far, the TRF circuit has probably seen the widest service. This is mainly true because it was used for many years in consumer radios.

One of the main problems with the TRF was the fact that each RF stage required individual tuning. This could be done either separately or with ganged capacitors. In either case it was troublesome. It would be much simpler if we could convert all the desired radio signals to one standard frequency. Then we could use *fixed tuned* amplifiers instead of tunable ones.

The *superheterodyne* receiver is a circuit which does work in this way. Figure 1-8 shows the block diagram of such a receiver.

In the superheterodyne receiver (frequently called simply a *superhet*) the incoming RF signal is combined with the output of the oscillator stage to produce an *intermediate frequency* signal. This intermediate frequency (or IF frequency) signal has been standardized for various types of service.

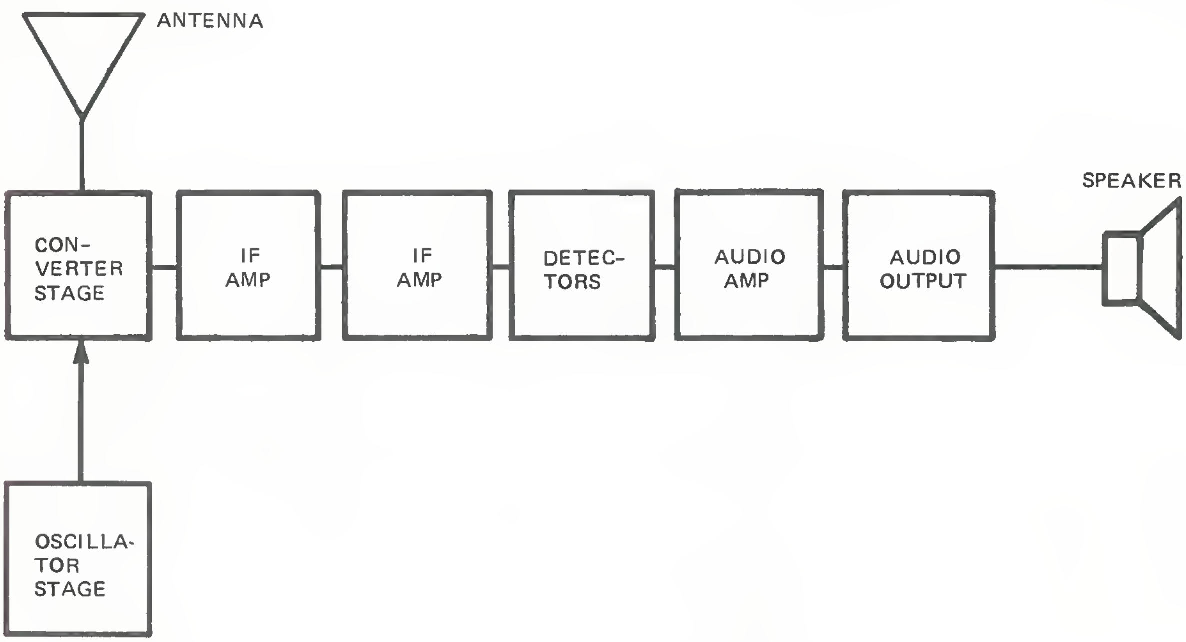


Fig. 1-8 Block diagram of a Superheterodyne Receiver

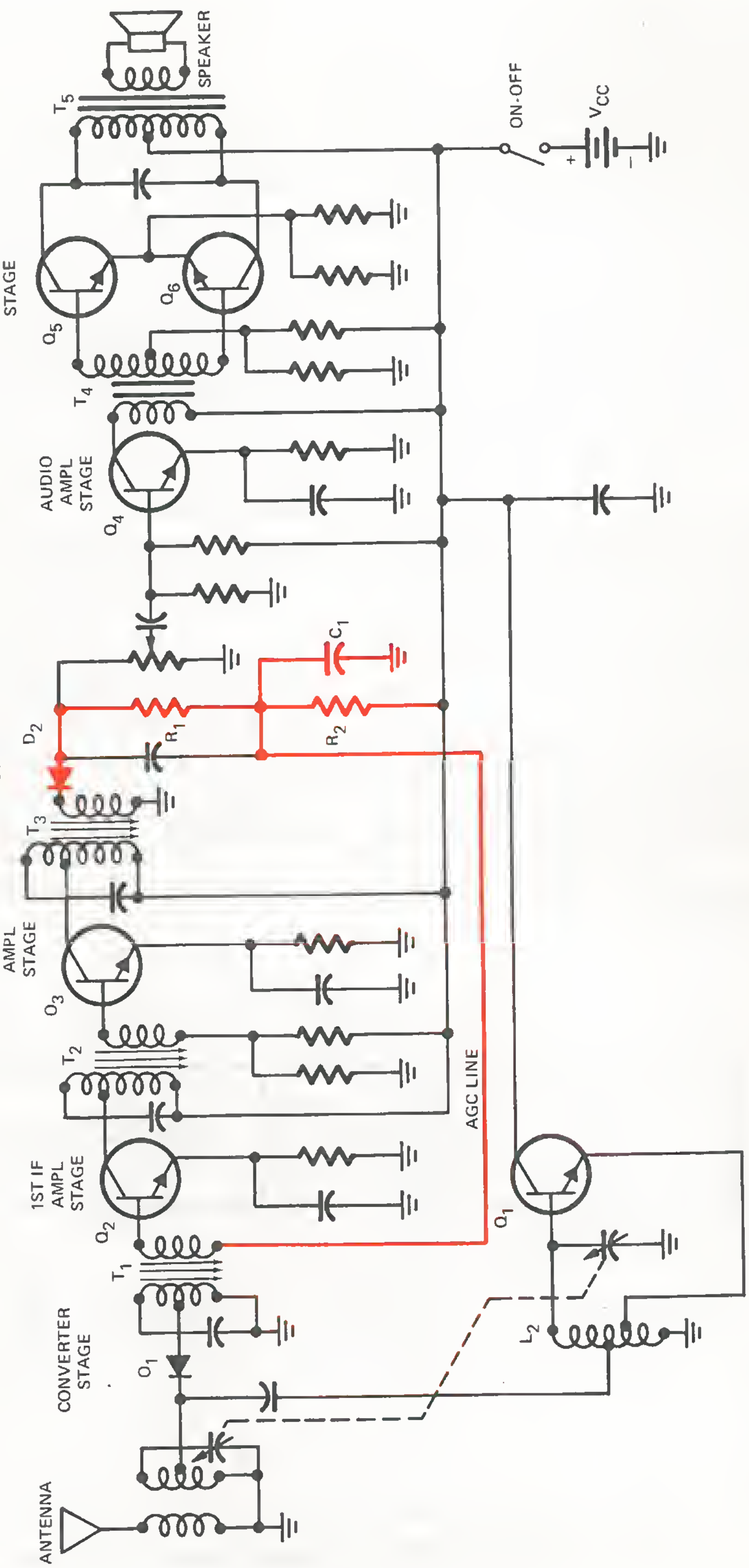


Fig. 1-9 A Superheterodyne Receiver Circuit

Here are a few of the more commonly encountered IF frequencies:

(a)	AM broadcast radio IF	455 kHz
(b)	FM broadcast radio IF	10.4 MHz
(c)	TV picture IF (Video sig.)	45.75 MHz
(d)	TV picture IF (Sound sig.)	41.25 MHz
(e)	TV sound IF (Intercarrier)	4.5 MHz
(f)	Radar IF	30 & 60 MHz

The stage in which the conversion takes place is called a *converter* or *mixer* stage. In this stage the incoming RF signal and an oscillator signal are mixed in such a way that the output frequency (IF) is equal to the difference between the two inputs. That is

$$f_{IF} = |f_O - f_{RF}| \tag{1.1}$$

For example, if the IF frequency is 455 kHz and the incoming RF is at 1000 kHz, then the oscillator is tuned to 1455 kHz.

The IF amplifier stages are fixed tuned to the IF (455 kHz in this case), and need not be retuned as the receiver is tuned from station to station.

The overall result then is that only the oscillator and antenna need to be tuned to select various incoming signals. Therefore, you will often find a two-gang tuning capacitor in a superhet receiver.

Some superhets have an RF amplifier between the antenna and converter stage. In a receiver of this type the RF amplifier will also require tuning.

Figure 1-9 shows the circuit diagram of a typical superheterodyne receiver. In this receiver a diode (D_1) is used for a converter. Some circuits use a single transistor for both the converter and the oscillator.

There is one important feature included in figure 1-9 that we have not discussed. That is *automatic gain control* (AGC, it is also sometimes called *automatic volume control,* AVC). Notice that the base bias for the first IF amplifier is produced by the voltage divider R_1 and R_2 across the capacitor C_1. One end of the divider is connected to the collector supply V_{CC}. The other end of the divider is connected to the detector output.

As the *average* output of the detector changes so does the bias on the first IF amplifier. In this way the gain of the IF amplifier is varied as the signal level varies. With a strong signal less gain is needed. Using AGC allows the output level of the receiver to remain relatively constant as the set is tuned from one station to another. AGC (or AVC) is used in almost all types of superhet receivers.

In some receivers the AGC is arranged so that it does not reduce the gain until the signal gets up to some preset value. This prevents the gain from being reduced even slightly on very weak signals. Such an AGC system is called *delayed automatic gain control* or DAGC (sometimes DAVC).

MATERIALS

1 AM radio receiver
1 Circuit diagram for the above receiver
1 Oscilloscope
1 RF signal generator
1 VOM or FEM

PROCEDURE

1. Examine the receiver diagram and determine what type of circuit it is. Record your decision on the data sheet.
2. Turn on the RF signal generator and oscilloscope. Allow them several minutes to warm up.
3. Set the signal generator for maximum output at a frequency of about 1 MHz (Mo. RF).
4. With the oscilloscope view the signal generator output waveform. Make a sketch of it on the data sheet and record the approximate carrier frequency, f_{RF}. (The waveform should be similar to figure 1-1c).
5. Turn on the receiver and loosely couple the signal generator to it. This can be done by connecting the signal generator to about one foot of unshielded wire. (A clip lead will do). Then lay the unshielded wire across the receiver's antenna coil.
6. Tune the receiver so that you can hear the generator test tone in the speaker.
7. Connect the oscilloscope leads across the oscillator section of the receiver tuning capacitor. Sketch the waveform and record the approximate oscillator frequency, f_O. (The waveform should be similar to figure 1-1b.) Use a X10 probe and retune the oscillator if necessary to hear the test tone.
8. Vary the receiver tuning and observe the oscillator waveform. Then retune the receiver as in step 6.
9. Connect the oscilloscope leads across the secondary of the last IF transformer. Sketch the waveform (It should be similar to figure 1-1c), and record the approximate value of the IF frequency, f_{IF}.
10. Connect the oscilloscope leads across the receiver volume control. Sketch the waveform (it should be similar to figure 1-1a) and record the approximate frequency of the signal, f_m.
11. Connect the oscilloscope across the speaker and compare the waveform to the one recorded in step 10.
12. Connect the VOM to measure the base-to-ground voltage of the first IF amplifier.
13. Record the base voltage for both maximum signal generator output and minimum signal generator output.
14. Connect the VOM to measure the base-to-ground voltage of the second IF amplifier.

15. Vary the signal generator output level and record the base voltage at the maximum and minimum generator settings.

16. Draw the block diagram of your receiver and identify the points at which each waveform in the data sheet occurs.

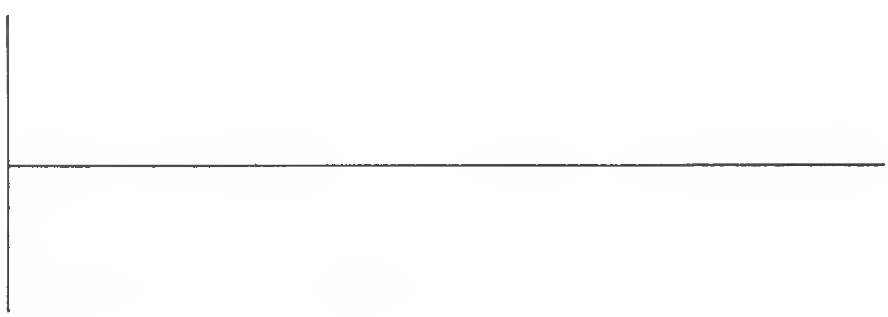
ANALYSIS GUIDE. In the analysis of your results you should explain what each waveform observed was. Identify the type of modulation and tell which portions of the waveform correspond to the RF carrier and to the modulation. Discuss the extent to which your frequency measurements verify the relationship given in equation 1.1.

Explain why the base voltage of the first IF stage varied with the signal generator output while that of the second IF stage did not.

Type of Receiver Circuit: ______________________________

Generator Output Waveform

$f_{RF} \approx$ ____________________



Oscillator Waveform

$f_0 \approx$ ____________________

Fig. 1-10 The Results

IF Amplifier Waveform

$f_{IF} \approx$ ______________________

Detector Output Waveform

$f_m \approx$

Audio Output Waveform

$f_m \approx$

	Maximum Gen Output	Minimum Gen Output
1st IF		
2nd IF		

Base Voltages

Fig. 1-10 The Results (Cont'd)

Fig. 1-11 Receiver Block Diagram

PROBLEMS

1. What are the four basic receiver operations?
2. What is receiver sensitivity?
3. What is receiver selectivity?
4. Does AGC affect selectivity or sensitivity? Explain why you think so.
5. A certain receiver has an IF frequency of 455 kHz. It is receiving a 1425 kHz signal. What is the oscillator frequency?
6. What would happen in problem 5 if there was another station operating at 2335 kHz? NOTE: Compare the difference between this frequency and the oscillator frequency to the 455 kHz IF frequency.
7. A radar receiver has a 60 MHz IF frequency and the oscillator operates at 9450 MHz. What is the frequency at which the radar operates?

8. Explain each of the following terms:

   (a) TRF receiver
   (b) Reflex receiver
   (c) Superhet
   (d) Regenerative receiver
   (e) Mixer
   (f) Detector
   (g) IF amplifier
   (h) RF amplifier

experiment 2 MODULATION AMPLIFIERS

INTRODUCTION. Most radio receivers have some type of modulation amplifier after the detector. In this experiment we shall examine the operating characteristics of some of the most common amplifiers.

DISCUSSION. The detector stage of a radio receiver separates the modulation signal from the RF carrier. In most cases, the recovered modulation is at such a low level that it cannot be used directly. As a result, it is normally necessary to amplify the modulation before it will perform the desired output function.

The modulation may be required to produce any of a great variety of output functions. For example, in the case of a broadcast radio the modulation must be able to drive the speaker. In a radar set, it must deflect a CRT beam, while in a petroleum pipeline link, it may have to operate an electric valve. In any case, the problem is to take the low-level detector output and amplify it to the required value.

There are four main considerations that come into play when we deal with a modulation amplifier. First, the gain of the amplifier must be appropriate for the particular application. Second, the frequency response of the amplifier must be such that the modulation can be satisfactorily reproduced. Third, the amplifier's input impedance must be high enough that it does not load down the detector too much for satisfactory operation. In some cases the detector and amplifier must have matched impedances. Fourth, the amplifier must have an output impedance that is low enough to drive the receiver load adequately. In many cases the amplifier output impedance must match the load impedance.

At this point let's look at a specific case and see how these considerations affect the actual circuitry. Let's suppose that we have a simple broadcast radio with the following parameters:

Detector output impedance ≈ 200 ohms
Maximum detector output ≈ 0.1 volts rms
Loudspeaker input impedance ≈ 8 ohms
Loudspeaker power capacity ≈ 250 mw

We can calculate the voltage that we should have across the speaker for maximum rated power

$$E = \sqrt{PR} = \sqrt{0.25 \times 8} \approx 1.4 \text{ volts rms}$$

Now since we know the maximum detector output and the maximum speaker input, we can compute the required voltage gain.

$$A_V = \frac{1.4}{0.1} = 14$$

That is, we need an amplifier with an *overall* voltage gain of about 14.

We see that since the detector output impedance is about 200 ohms, we will want our amplifier to have an input impedance considerably greater than 200 ohms. Since most common emitter amplifiers have an input impedance of about 1000 ohms or more, we would probably choose such a circuit.

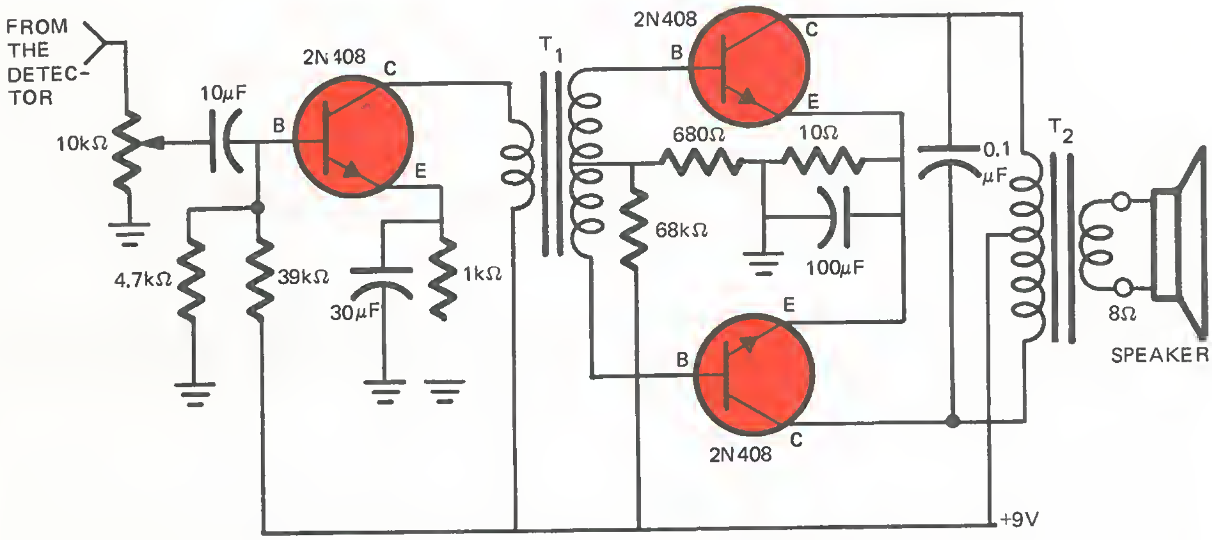


Fig. 2-1 A Typical Radio Audio Section

Figure 2-1 shows a circuit that is typical of broadcast radio audio sections.

Ordinarily, an AM broadcast station is limited to a bandwidth of 10 kHz. This being the case, the modulation bandwidth is from about 100 Hz to about 5000 Hz.

In FM radio, a modulation band from about 20 Hz to about 20 kHz is commonly used. As a result, the modulation amplifiers in an FM radio must have a much broader frequency response than those of an AM radio.

Modulation amplifiers for systems which handle pulses and video must have even broader bandwidths. A television receiver, for example, has video amplifiers that have bandwidths of several megahertz.

There are a number of ways in which the frequency response of an amplifier may be extended. Let's look at a couple of the simpler methods. First let's suppose that we wish to extend the low-frequency response of the first stage in figure 2-2. Since the low-frequency response is determined by

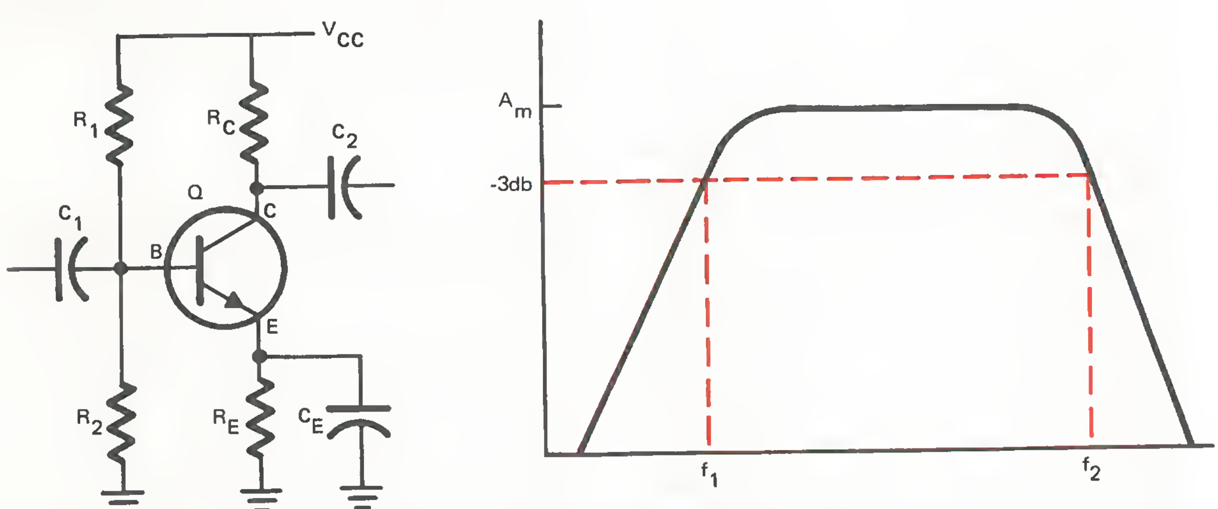


Fig. 2-2 An Uncompensated Amplifier

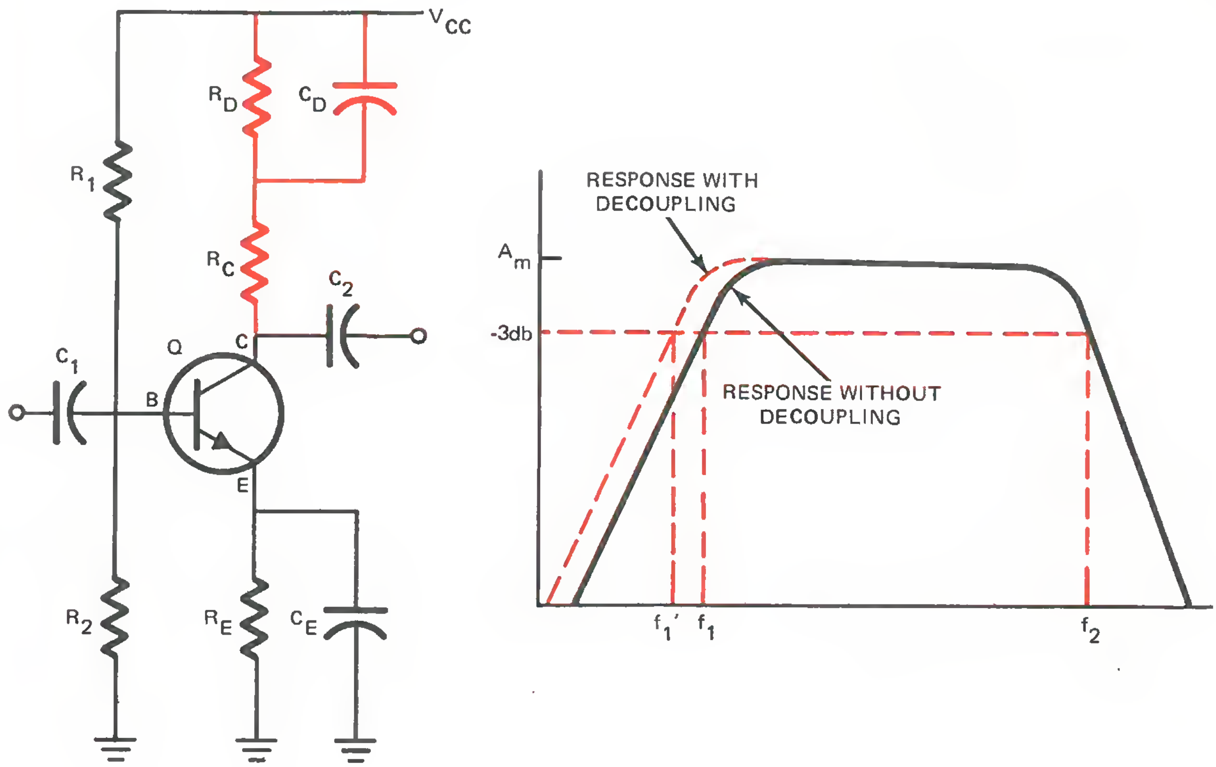


Fig. 2-3 Low-Frequency Decoupling Network

the interaction of the coupling capacitors and the other elements, we can extend the response by using larger capacitors. In a practical case we would choose the largest reasonable capacitor values that were available for C_1 and C_2.

We could also remove the emitter bypass capacitor. This would allow some emitter degeneration with a resulting loss of gain but an increase in bandwidth.

We could also use direct coupling which would extend the frequency response downward to zero frequency.

A final method of extending low frequency response is to use a decoupling network in the collector circuit. This method is illustrated in figure 2-3. In this circuit the low-frequency decoupling network, R_D and C_D, have little effect at the mid frequencies and above because C_D effectively bypasses R_D. At low frequencies the bypassing effect of C_D decreases, which causes an increase in the total collector circuit resistance. As the collector circuit resistance goes up, so does the stage gain. The overall result is a lowering of f_1 to the value f_1'.

The value of R_D is usually about 10 percent of R_C and C_D is chosen so that

$$\boxed{C_D = \frac{1}{2\pi f_1 R_D}} \qquad (2.1)$$

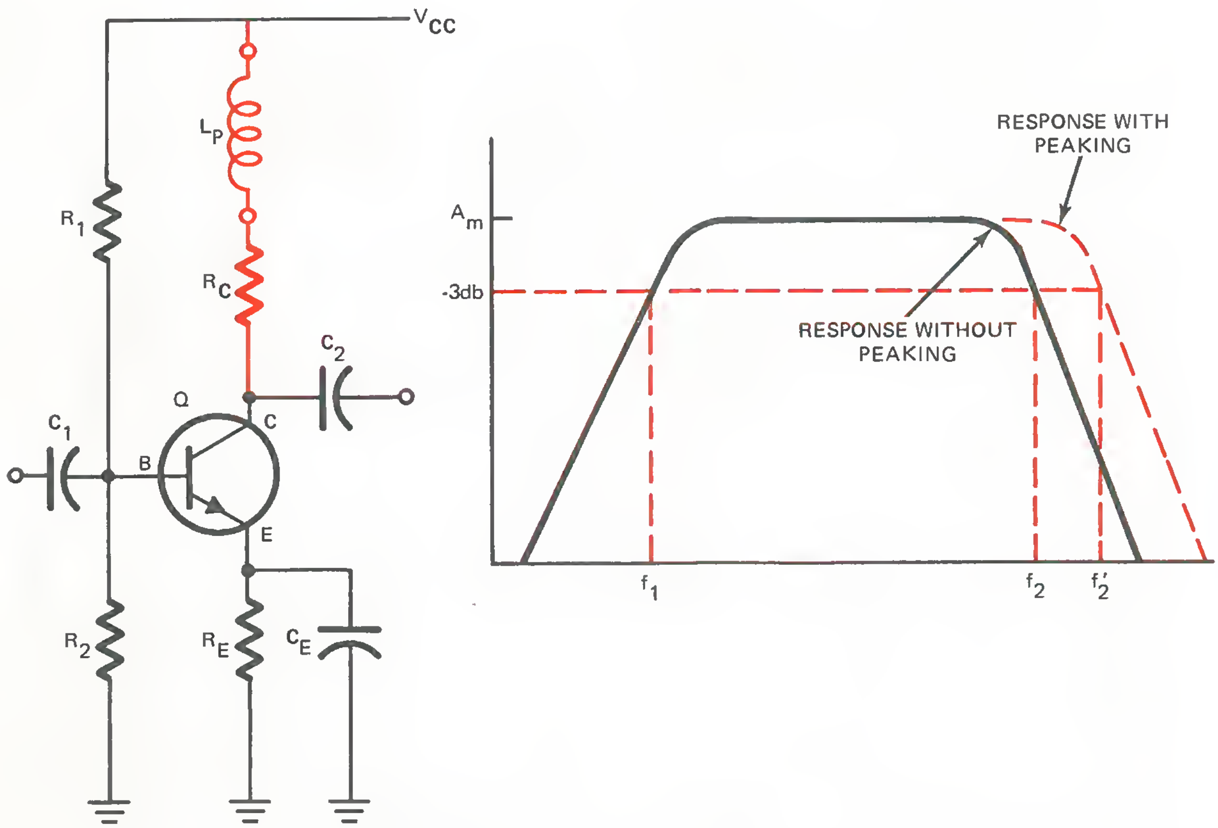


Fig. 2-4 Shunt Peaking An Amplifier

This type of low-frequency compensation is often used in wideband modulation amplifiers.

High-frequency compensation is done in somewhat the same manner. Since the decrease in gain at the high-frequency end of the response is caused primarily by shunt capacitance, the problem is to cancel out the effect of this shunt capacitance. Figure 2-4 shows our amplifier circuit with high-frequency compensation. This type of compensation is called *shunt peaking* and the inductance L_p is called a *peaking coil.* The value of L_p is chosen such that its effect on the gain just balances that of the effective shunt capacitance of the circuit. The usual L_p value is

$$\boxed{L_p = \frac{R_C}{2\pi f_2}} \tag{2.2}$$

Series Peaking is also sometimes used for high-frequency compensation. In this case, the series peaking coil is connected in series with the coupling capacitor C_2.

Most broadband video and pulse amplifiers employ both high- and low-frequency compensation. Figure 2-5 shows a circuit with both types of compensation. Notice that in this case the decoupling capacitor C_D could be connected from one side of R_D to the ground. This would produce the same effect as placing it right across R_D.

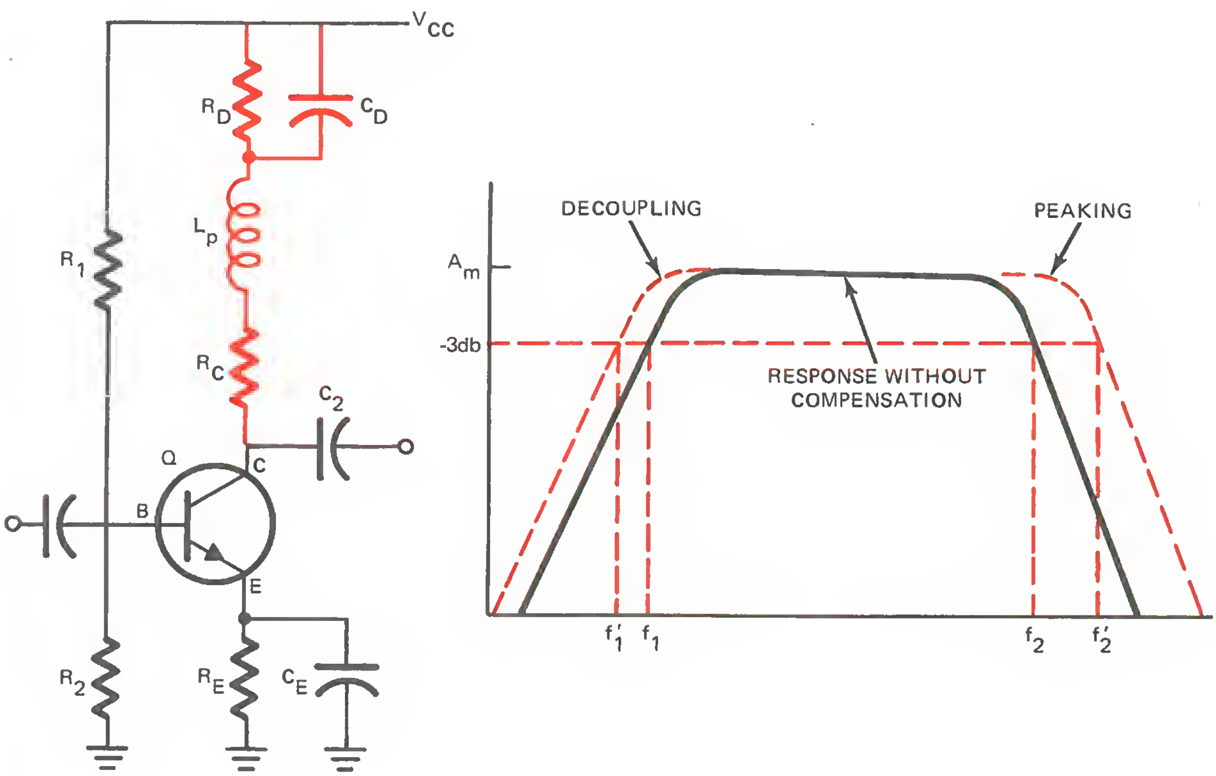


Fig. 2-5 Amplifier with Decoupling and Peaking

MATERIALS

- 1 AM radio receiver
- 1 Schematic diagram for the receiver
- 1 Audio signal generator
- 1 Oscilloscope
- 1 Sheet of 4-cycle, semilog graph paper
- 1 VOM or FEM

PROCEDURE

1. Examine the receiver diagram. Record the type and component values of any compensation networks in the audio section.
2. Connect the audio generator across the volume control of the receiver. Set the generator for minimum output at 1000 Hz.
3. Connect the oscilloscope across the speaker terminals.
4. Connect the VOM to read the signal voltage across the volume control.
5. Set the receiver volume control for minimum volume. Then increase the signal generator output level until the VOM reads about 1/2 volt rms.
6. Now slowly increase the receiver volume until you observe some flattening of the peaks in the speaker signal. Reduce the volume until *no* flattening at all is observed.

7. Do not change the volume control setting again until you are instructed to do so.
8. Record the peak-to-peak speaker voltage for each frequency shown in the data table. At each step adjust the signal generator output level if necessary to keep the VOM reading the same as step 5.
9. If there is a compensation network in your audio section, remove it *and* the emitter bypass capacitors. Then repeat steps 5 through 8. If there is no compensation network, remove only the emitter bypass capacitor from each amplifier stage. Then repeat steps 5 through 8.
10. On a single sheet of 4-cycle semilog graph paper plot output voltage (vertically) versus frequency (horizontally) for each of your two data runs.
11. Replace any components that you removed from the receiver.

ANALYSIS GUIDE. In your analysis of these data you should compare your two frequency response curves. Which one has the wider bandwidth? Why is this so? Explain why the first one is better suited to the receiver operation than the second one.

Type and Value of

Compensation Components ______________________________

____

FREQ	FIRST VOLTAGE	SECOND VOLTAGE	FREQ	FIRST VOLTAGE	SECOND VOLTAGE
50 Hz			6,000 Hz		
100 Hz			8,000 Hz		
200 Hz			10,000 Hz		
500 Hz			12,000 Hz		
1000 Hz			14,000 Hz		
2000 Hz			16,000 Hz		
4000 Hz			18,000 Hz		

Fig. 2-6 The Data Table

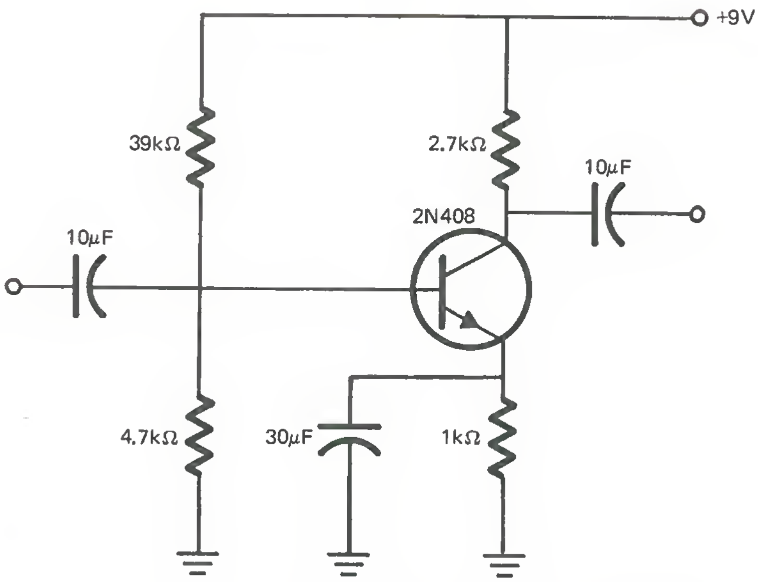


Fig. 2-7 Circuit For Problem 1

PROBLEMS

1. The amplifier circuit shown in figure 2-7 has a midband voltage gain of 50. The value of f_2 is found to be 37.5 kHz while f_1 is 170 Hz. Draw a new circuit diagram showing the use of both decoupling and shunt peaking. Give the values of R_D, C_D, and L_p.

2. The circuit diagram in figure 2-5 has the following component values:

$R_1 = 27\ k\Omega$	$C_E = 100\ \mu F$	$C_D = 10\ \mu F$
$R_2 = 4.7\ k\Omega$	$R_E = 680\Omega$	$R_D = 470\Omega$
$C_1 = 5\ \mu F$	$R_C = 3.9\ k\Omega$	Q = 2N408
$C_2 = 5\ \mu F$	$L_p = 15\ mH$	$V_{CC} = 9V$

   Assuming that equations 2.1 and 2.2 hold, what were the values of f_1 and f_2 before compensation?

3. Explain each of the following terms:

   (a) peaking
   (b) amplifier bandwidth
   (c) compensation
   (d) decoupling
   (e) modulation amplifier

experiment 3 AM DETECTORS

INTRODUCTION. Detection is one of the basic functions that every receiver must provide. In this experiment we shall examine some of the circuits used to perform this operation.

DISCUSSION. Perhaps the simplest AM detector circuit that is used in a practical application is the diode arrangement shown in figure 3-1. In this circuit the modulated IF signal is coupled into the detector from the left. The diode is the actual detecting device. The capacitor C is the detector filter which separates the modulation from the IF carrier. R is the effective load on the detector.

The detector diode itself has a nonlinear characteristic such as the one shown in figure 3-2a. The diode characteristic curve can be described mathematically in a number of ways. One of these ways is by use of a power series such as

$$i = a_0 + a_1 e_d + a_2 e_d^2 + a_3 e_d^3 + \dots + a_n e_d^n$$

The whole detector circuit having a similar

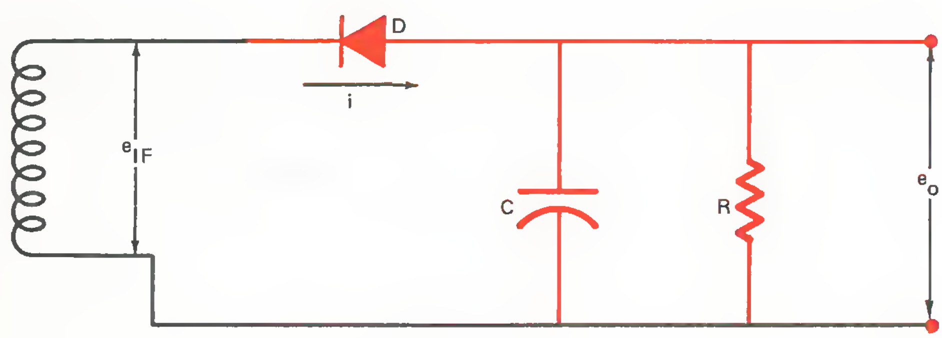


Fig. 3-1 A Diode Detector

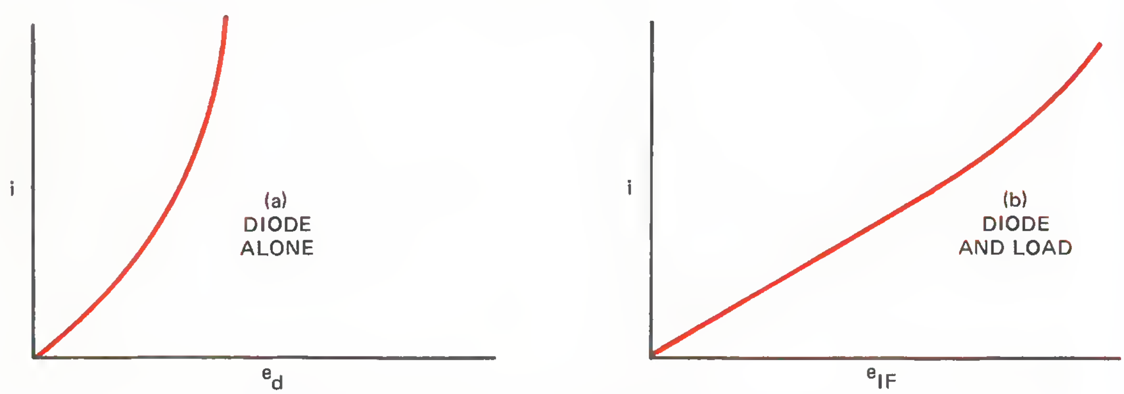


Fig. 3-2 Characteristic Curves of Diode and Circuit

characteristic can be described by a similar series

$$i = A_o + A_1 e_{IF} + A_2 e_{IF}^2 + A_3 e_{IF}^3 + \ldots + A_n e_{IF}^n$$

The first term in this series (A_o) is the bias current, if any, through the circuit. In most diode detectors the value of A_o will be zero. For *very, very* small input voltages the term ($A_2 e_{IF}^2$) becomes dominant. In this region the detector is called a *square law detector* and the equation

$$\boxed{i \cong A_2 e_{IF}^2} \qquad (3.1)$$

is a reasonable approximation of the circuit performance. Most detectors work in this square law region.

If the input voltage is relatively large, the fourth, fifth, sixth, and on terms of the power series become important. In such a case the detector is no longer operating in the square law region. Very few receiver detectors have such large input signals, so we shall concentrate on square law operation only.

The IF signal that is the input for the detector has two main components, the IF carrier and the modulation. An amplitude modulated IF signal can be described by the trigonometric equation

$$e_{IF} = E_{IF} \sin \omega_{IF} t + \frac{1}{2} E_m \cos (\omega_{IF} t + \omega_m t) + \frac{1}{2} E_m \cos (\omega_{IF} t - \omega_m t) \qquad (3.2)$$

While this equation is somewhat lengthy, it does have three readily identifiable parts. The first term ($E_{IF} \sin \omega_{IF} t$) is the IF carrier signal. The second and third terms are the modulation components and are called the *upper and lower sidebands* respectively. Notice that each sideband differs in frequency from the IF carrier by an amount equal to the modulation frequency. Also notice that each sideband has an amplitude equal to half the modulation amplitude.

If we substitute this long trigonometric relationship into the square law equation (3.1) we have

$$i = A_2 \left[E_{IF} \sin \omega_{IF} t + \frac{1}{2} E_m \cos (\omega_{IF} t + \omega_m t) + \frac{1}{2} E_m \cos (\omega_{IF} t - \omega_m t)\right]^2$$

Squaring the trigonometric polynomial renders an even more involved relationship. However, by substituting the trig identities,

$$\sin^2 \omega t = \frac{1}{2} (1 - \cos 2\omega t),$$

$$\cos^2 \omega t = \frac{1}{2} (1 + \cos 2\omega t),$$

$$\sin \omega_1 t \cos \omega_2 t = \frac{1}{2} [\sin (\omega_1 t + \omega_2 t) + \sin (\omega_1 t - \omega_2 t)]$$

and

$$\cos \omega_1 t \cos \omega_2 t = \frac{1}{2} [\cos (\omega_1 t + \omega_2 t) + \cos (\omega_1 t - \omega_2 t)]$$

we can arrive at an equation for the detector current,

$$i = \left.A_2 E_{IF}^2 + \frac{A_2}{4} E_m^2 \right\} \text{DC Terms}$$

$$\left.\begin{array}{l} - A_2 E_{IF}^2 \cos 2\omega_{IF} t \\ - A_2 E_m^2 \cos 2\omega_{IF} t \end{array}\right\} \text{High Frequency Terms}$$

$$\left.\begin{aligned}&+\frac{A_2}{4}E_m^2\cos 2(\omega_{IF}t-\omega_m t)\\&+\frac{A_2}{4}E_m^2\cos 2(\omega_{IF}t+\omega_m t)\\&+\frac{A_2}{2}E_{IF}E_m\sin(2\omega_{IF}t-\omega_m t)\\&-\frac{A_2}{2}E_{IF}E_m\sin(2\omega_{IF}t+\omega_m t)\end{aligned}\right\}\begin{array}{l}\text{High}\\\text{Freq.}\\\text{Terms}\end{array}$$

$$+A_2E_{IF}E_m\sin\omega_m t\Big\}\ \text{Modulation}$$

$$-\frac{A_2}{4}E_m^2\cos 2\omega_m t\Big\}\ \text{Distortion}$$

Of this ten-term expression, the six high-frequency terms are filtered out by the detector filter capacitor. The remaining terms constitute the current flow through the detector load. In other words, the load current is

$$i_L=\underbrace{A_2(E_{IF}^2+\tfrac{1}{4}E_m^2)}_{\text{Direct Current}}+\underbrace{A_2E_{IF}E_m\sin\omega_m t}_{\text{Modulation Signal}}$$

$$\underbrace{-\frac{A_2}{4}E_m^2\cos 2\omega_m t}_{\text{Distortion}}$$

As we see there are three load current components, each of which is affected by A_2 and E_{IF}. A_2 is the detector characteristic coefficient. E_{IF} is the peak value of the IF carrier signal. All three components are affected by E_m, the peak modulation amplitude.

You should notice that there is always some distortion present at the output of a square law detector. However, most detector designs are such that this distortion is relatively small. The distortion can be neglected if $1/4E_m^2$ is considerably smaller than E_{IF}. When this is the case, the detector load current becomes approximately

$$i_L\approx A_2E_{IF}+A_2E_{IF}E_m\sin\omega_m t \qquad (3.3)$$

This expression has two terms: a direct current term and a modulation term. The direct current term, which is proportional to the IF carrier amplitude, is used for automatic gain control of the IF amplifiers. The modulation signal is coupled to the modulation amplifier section of the receiver.

Figure 3-3 shows a detector circuit with the AGC and modulation take off points.

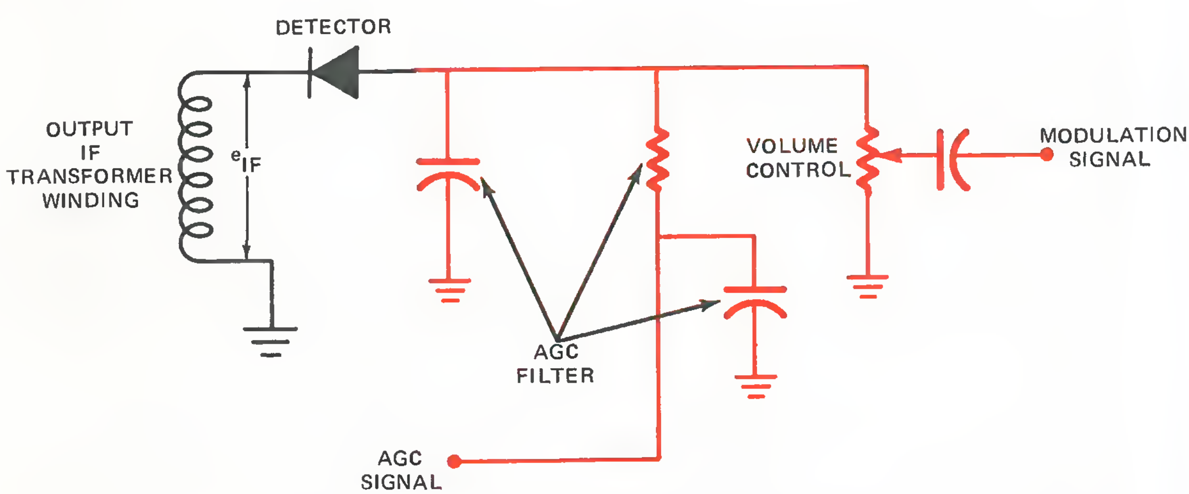


Fig. 3-3 Detector with Output Points

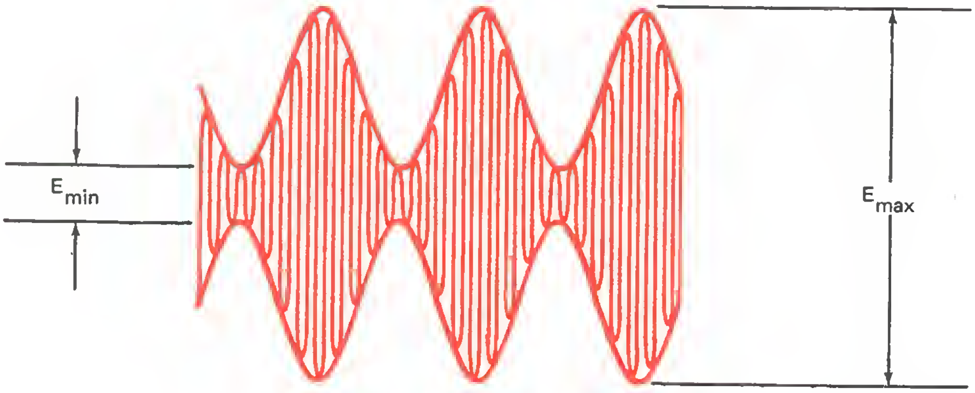


Fig. 3-4 A Modulated IF Signal

We can measure both the AGC voltage (using a VOM) and the modulation signal (using an oscilloscope) at the output of the detector.

Measuring the values of E_{IF} and E_m from the modulated IF signal is somewhat more involved. It can, however, be done as shown in figure 3-4 by measuring E_{max} and E_{min}. Based on equation 3.2, the maximum possible peak-to-peak amplitude of this waveform is

$$E_{max} = 2E_{IF} + 2E_m \qquad (3.4)$$

while the minimum possible value is

$$E_{min} = 2E_{IF} - 2E_m \qquad (3.5)$$

provided that E_{IF} is equal to or greater than E_m. When E_m is greater than E_{IF}, considerable distortion occurs. This condition is normally avoided in receivers handling sinusoidal modulation signals.

Solving equations 3.4 and 3.5 simultaneously we see that

$$E_{max} + E_{min} = 4E_{IF}$$

or

$$\boxed{E_{IF} = \frac{1}{4}(E_{max} + E_{min})} \qquad (3.6)$$

Also

$$E_{max} - E_{min} = 4E_m$$

or

$$\boxed{E_m = \frac{1}{4}(E_{max} - E_{min})} \qquad (3.7)$$

Consequently, we can determine E_{IF} and E_m by measuring E_{max} and E_{min} on the modulated IF signal.

Some receivers use a transistor detector circuit instead of a diode. Such a circuit is often called a *power detector.* Figure 3-5 shows a typical power detector circuit.

As you will recall, the input characteristic of a transistor is substantially the same as that of a diode (see figure 3-2a). As a result, the power detector works very much like a diode circuit except that we have a current gain (h_{fe}) from base to collector. Therefore, the detector characteristic coefficient, A_2, in the previous equations is much larger for the transistor than for a diode.

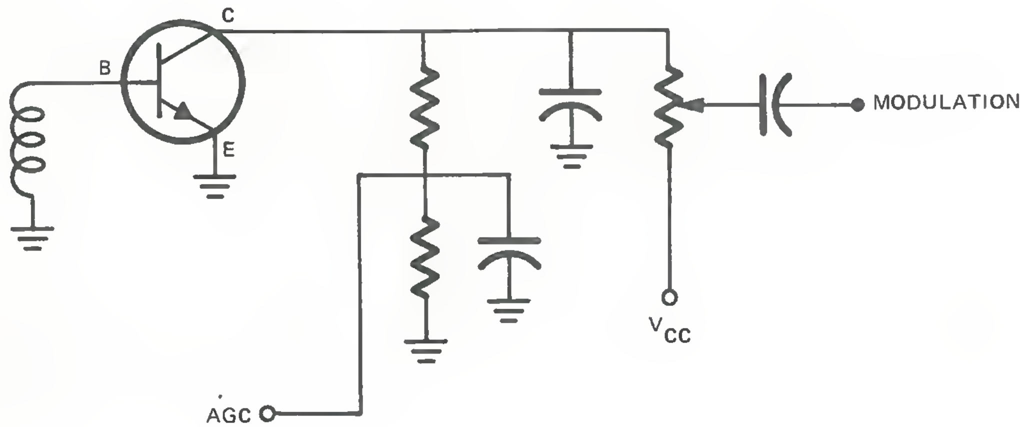


Fig. 3-5 A Power Detector Circuit

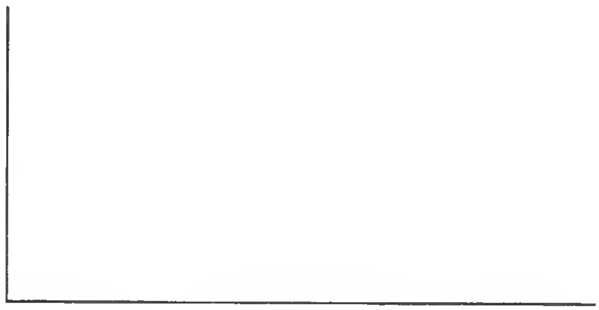
MATERIALS

1 AM radio receiver
1 Schematic diagram for the receiver
1 RF signal generator
1 Oscilloscope
1 VOM or FEM

PROCEDURE

1. Connect all of the equipment to the appropriate power sources and allow time for warm up.
2. Loosely couple the RF signal generator to the receiver's antenna. This can be done by connecting a foot or so of unshielded wire (a clip lead will do) to the generator cable and and looping it over the receiver's antenna coil.
3. Tune the generator for about 1.0 MHz and set it for maximum output.
4. Tune the receiver for the maximum signal generator test tone output. Set the receiver volume to a comfortably low output level.
5. Connect the oscilloscope across the detector input winding of the last IF transformer.
6. Adjust the scope for viewing the modulated IF carrier signal. Retune the receiver if necessary to produce the maximum IF output.
7. Make a sketch of the IF waveform and record the values of E_{max} and E_{min}.
8. Calculate and record the values of E_{IF} and E_m.
9. Move the oscilloscope to the output side of the detector diode.
10. Make a sketch of the output modulation waveform and record its *peak* value E_0.

11. Connect the VOM from the detector output to the ground.
12. Vary the signal generator output level from zero to maximum and record the change, ΔE_{DC}, in DC detector voltage.
13. From the circuit diagram determine the value of the *DC load resistance,* R_{DC}, on the detector.
14. Compute the DC detector current using $I_{DC} = \Delta E_{DC}/R_{DC}$.
15. With the DC detector current and equation 3.3 (use only the DC term) compute A_2 ($I_{DC} = A_2 E_{IF}$).
16. From the circuit diagram determine the value of the modulation load current I_0, using $I_0 = E_0/R_m$.
17. With the modulation term from equation 3.3 compute the *peak* output modulation current I'_0 (use $I'_0 = A_2 E_{IF} E_m$).

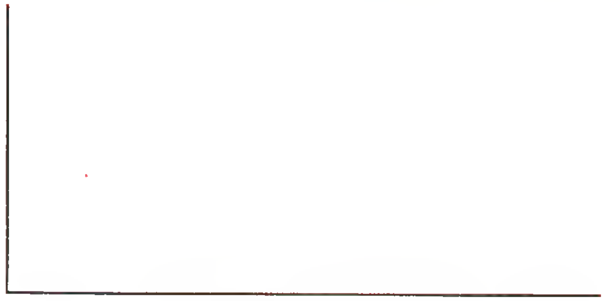


E_{max} = ____________

E_{min} = ____________

IF Waveform

E_{IF}	E_m	E_{dc}	R_{dc}	I_{dc}	A_2	R_m	I_0	I'_0



E_0 = ____________

Modulation Signal

Fig. 3-6 The Data Tables

ANALYSIS GUIDE. In the analysis of these data you should compare your values of I_0 and I'_0 to each other. Discuss the differences in the two values. Also discuss the extent to which your results agreed with the material presented in the discussion section of the experiment.

PROBLEMS

1. Discuss in your own words how a diode detector works. How is it different from a rectifier?

2. Define each of the following terms in your own words.
   (a) Upper sideband
   (b) IF carrier
   (c) AGC filter
   (d) Power detector
   (e) Square law detector
   (f) Detector characteristic coefficient

3. A diode detector has a modulation load of 20 kΩ and a DC load of 8 kΩ. The detector characteristic coefficient is equal to 10^{-9} and the input signals are $E_{IF} = 0.5$ volts, $E_m = 0.1$ volts. What is the value of the peak modulation output voltage?

4. What is the value of the AGC voltage in problem 3?

5. A certain detector circuit has an input of

$$E_{IF} = 0.6 \sin 2.86 \times 10^6 t + 0.06 \cos 2866280t + 0.06 \cos 2853720t$$

   Determine each of the following quantities:
   (a) The peak IF carrier voltage
   (b) The peak modulation voltage
   (c) The IF frequency
   (d) The modulation frequency

6. What would have been the results in problems 3 and 4 if the signal in problem 5 had been used?

experiment 4 *IF AMPLIFIERS*

INTRODUCTION. Intermediate frequency amplifiers are used in virtually all superheterodyne receivers. In this experiment we shall examine the operating characteristics of tuned amplifiers like those frequently used as IF amplifiers.

DISCUSSION. IF amplifiers are usually transformer-coupled tuned amplifiers. Figure 4-1 shows a typical IF amplifier circuit. Notice that all of the stages are more or less identical common emitter amplifiers. Figure 4-2 shows one of these stages by itself. The only real differences between this circuit and an audio amplifier are the tuned circuit and the values of the bypass capacitors. The bypass capacitors are much smaller in value than would be the case in an audio circuit. However, all that is necessary is that the reactance of the capacitors be much smaller than the resistors they are bypassing at the IF frequency. If the IF

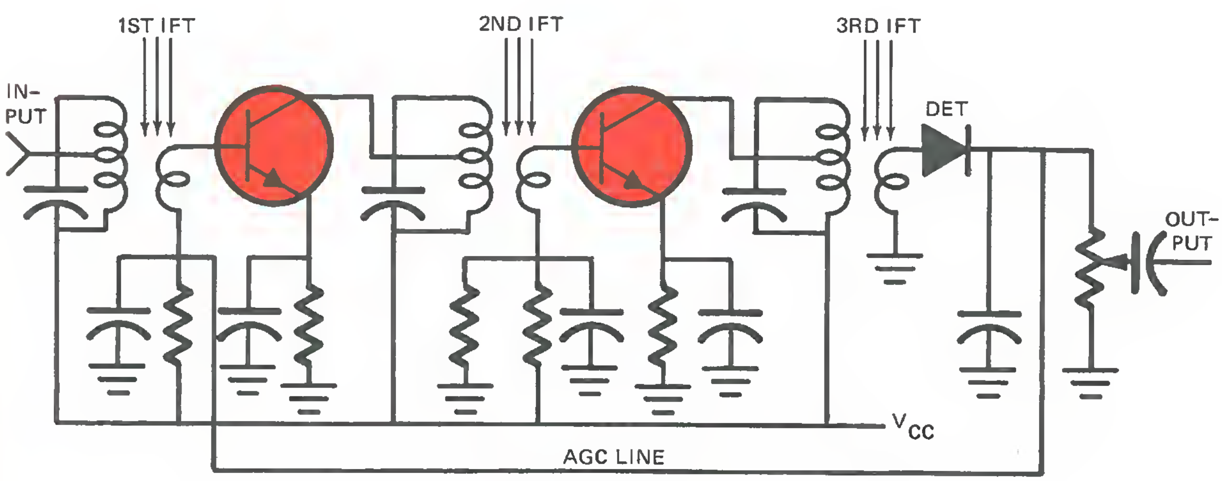


Fig. 4-1 An IF Amplifier Circuit

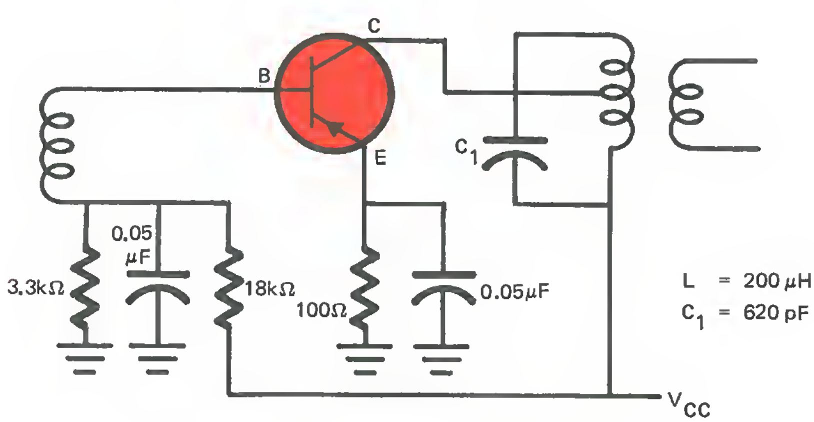


Fig. 4-2 A Single Stage IF Amplifier

frequency is 455 kHz then the reactance of the bypass capacitors is

$$X_C = \frac{1}{2\pi f_C} = \frac{1}{6.28 \times 455 \times 10^3 \times 0.05 \times 10^{-6}}$$

or

$$X_C = 7 \text{ ohms at } 455 \text{ kHz}$$

Notice that this is much less than either of the two bypassed resistors. The 0.05 μF capacitors are, therefore, quite satisfactory for use as bypass capacitors in this circuit.

The tuned circuit provides the collector circuit load for the stage. Provided that the Q of the circuit is relatively high, we can calculate the resonant frequency using

$$f_r \approx \frac{1}{2\pi\sqrt{LC}} = \frac{1}{6.28\sqrt{200 \times 620 \times 10^{-18}}}$$

$$f_r \approx \frac{0.159}{3.5\ 10^{-7}} = 455 \text{ kHz}$$

The voltage gain of a common emitter amplifier is

$$A_V = \frac{-h_{fe}R_L}{h_{ie} + \Delta_h R_L} \qquad (4.1)$$

where

$$\Delta_h = h_{ie}h_{oe} - h_{fe}h_{re}$$

An examination of equation 4.1 reveals that the voltage gain is more or less directly proportional to the effective collector load R_L. In our IF amplifier the tuned circuit is the collector load.

The impedance of the tuned circuit will vary with frequency pretty much as shown in figure 4-3. Since the amplified gain varies with the collector load value, this figure is also the frequency response of the amplifier.

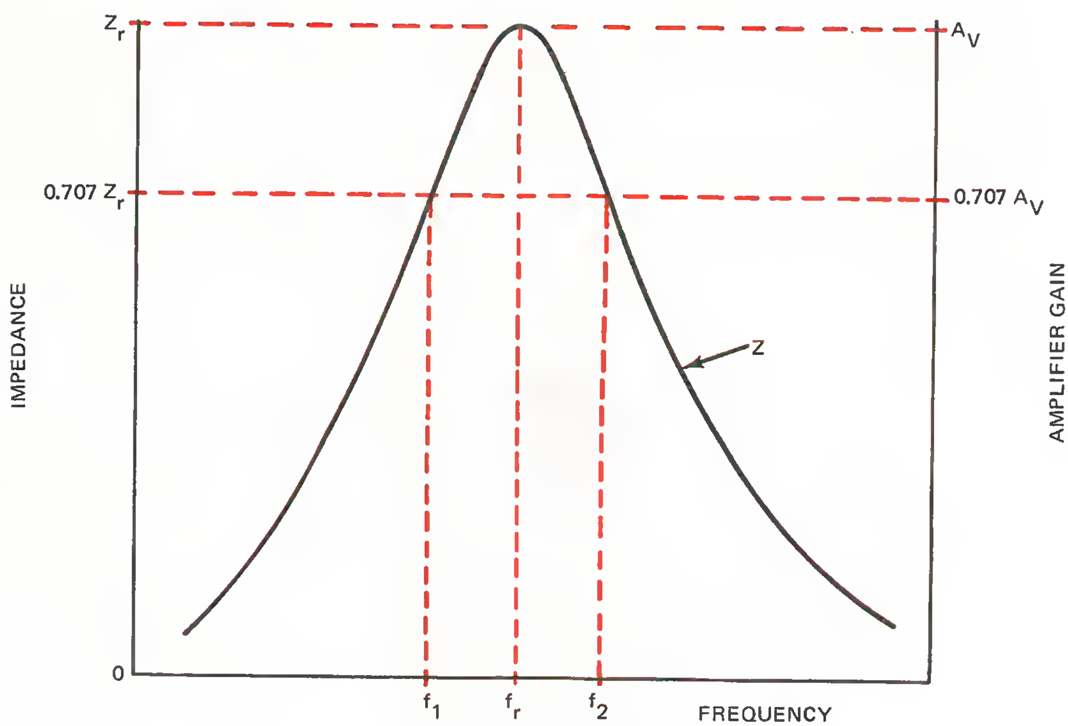


Fig. 4-3 Impedance of a Tuned Circuit

The bandwidth of our single stage amplifier will be the same as the tuned circuit bandwidth

$$\boxed{BW = \frac{f_r}{Q}} \tag{4.2}$$

For a commercial broadcast AM radio the bandwidth must be 10 kHz since this is the channel width of the station. So, when we set the IF frequency and the channel width, we have also set the Q that the tuned circuit must have.

If we add another stage of IF amplification the overall bandwidth will be reduced. To understand why this happens suppose that each amplifier has a 10 kHz bandwidth and the IF frequency is 455 kHz. Now, when we come down to 450 kHz both amplifier gains are down to 0.707 times their center value. (This is a 3-db reduction in each stage gain.) The overall gain is now down to 0.707 A_{V1} X 0.707 A_{V2} = 0.5 $A_{V1}A_{V2}$ because we have to multiply the two gains to get their combined gain. The combined gain then is 6 db down from the center frequency. In other words, the overall bandwidth has been reduced by adding another stage. We can correct this by increasing the bandwidth of each stage. The bandwidth is increased if we reduce the Q of each tuned circuit. For n identical stages, the Q required for a given bandwidth and IF frequency is

$$Q = (2^{\frac{1}{n}} - 1)^{\frac{1}{n}} \frac{f_{IF}}{BW} \tag{4.3}$$

So, for a 455-kHz IF with a 10-kHz bandwidth and two IF amplifier stages we require a tuned circuit Q of

$$Q = (2^{\frac{1}{2}} - 1)^{\frac{1}{2}} \frac{455}{10} \approx 30$$

And the bandwidth of *each* stage will be

$$BW = \frac{f_r}{Q} = \frac{455}{30} = 15.2 \text{ kHz}$$

The amplifiers that we have considered so far have been *single tuned* circuits. That is, only one side of the IF transformer was tuned to the IF frequency. Such circuits are popular because they are economical and easy to work with. When flatter frequency response and steeper sides on the response curve are needed to reject adjacent channel signals *double* tuned transformers may be used. Figure 4-4 shows a doubled tuned IF amplifier circuit.

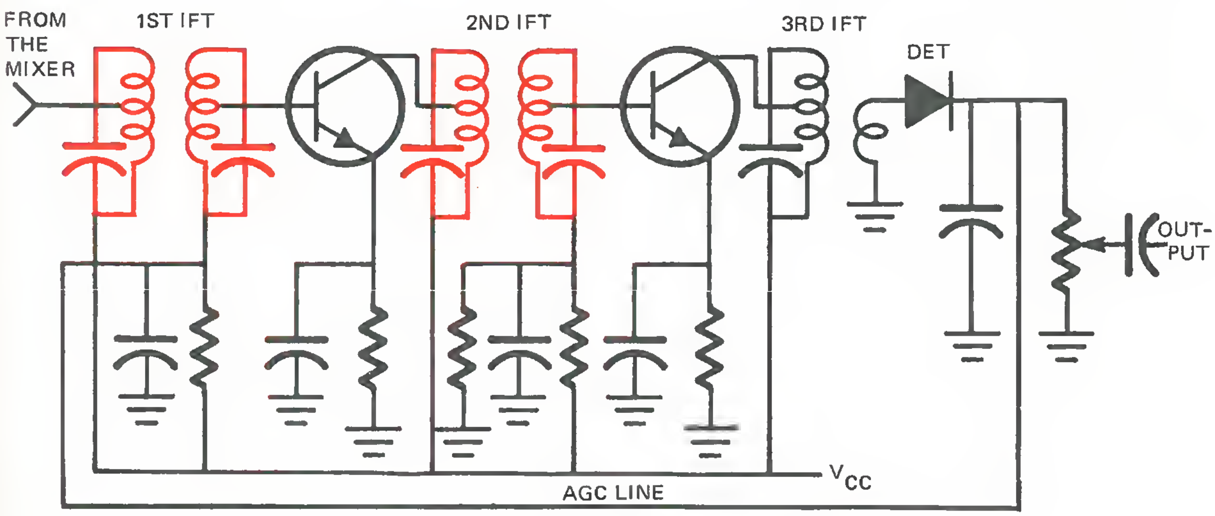


Fig. 4-4 A Double Tuned IF Amplifier

In order for any IF amplifier to work correctly, it must be properly tuned. The process of tuning is called *alignment* of the IF stages. Commercial radios are probably the easiest circuits to align because they are all tuned to the same frequency. All we have to do is connect an indicator of some kind to the detector output; then couple a signal into the set and adjust the oscillator for maximum output signal; then tune the last IF transformer for the maximum output indication. When this is done move back to the next IF transformer and again tune for maximum output indication. When each IF transformer has been adjusted for maximum output, start over with the last one and repeat the whole procedure again. Keep doing this until no further increase in output can be achieved. The IF amplifiers are now aligned.

MATERIALS

1 AM radio receiver
1 RF signal generator
1 Oscilloscope
1 Nonmetallic hex-type alignment tool
1 Circuit diagram for the receiver

PROCEDURE

1. Turn on all of your equipment and allow it to warm up.
2. Connect the oscilloscope across the detector output.
3. Set the receiver tuning capacitor to its maximum frequency.
4. Loosely couple the signal generator to the receiver antenna.
5. Set the signal generator frequency to 455 kHz. Set its output to maximum. You should now be able to hear the generator test tone at the speaker.
6. Reduce the signal generator output level until you can just hear the signal at the speaker with the receiver volume turned up full.
7. Readjust the generator frequency slightly if necessary to produce the maximum output, then repeat step 6.
8. With the nonmetallic alignment tool adjust the last IF transformer for maximum output.
9. Repeat step 6.
10. Now tune the next IF transformer for maximum output.
11. Repeat step 6.
12. Now adjust the first IF transformer for maximum output.
13. Repeat step 6.
14. Go back and do steps 8 through 13 over several times until you go through all of the steps without getting any increase in output.

15. Move the oscilloscope to the input of the detector stage. Connect it across the secondary of the last IF transformer.
16. Increase the generator output until you get the maximum *undistorted* output. Turn the receiver volume down to zero.
17. Record the peak-to-peak amplitude and the frequency of the signal (E_0 and f_{IF}).
18. Move the oscilloscope to the secondary of the second IF transformer.
19. Record the peak-to-peak voltage (E_i).
20. Compute the gain of the last IF amplifier stage.
21. Return the oscilloscope to the secondary of the last IF transformer.
22. Carefully reduce the generator frequency until the output voltage is 0.707 times the value recorded in step 17. Record the frequency (f_1).
23. Carefully increase the signal frequency until the output voltage goes through maximum and down to 0.707 times the value recorded in step 17. Record the frequency (f_2).
24. Compute the overall bandwidth of the IF section.
25. Move the signal generator and connect it across the secondary of the second IF transformer. Set it to the frequency recorded in step 17.
26. Repeat step 16. Then repeat steps 22 and 23. Record your results as f'_1 and f'_2.
27. Compute the bandwidth of the last IF amplifier stage.

ANALYSIS GUIDE. In analyzing your results from this experiment you should discuss the reason for doing each step in the alignment procedure (steps 2 through 14). Also explain why your two values of bandwidth do not agree.

PROBLEMS

1. What was the Q of the tuned circuit in the last IF stage?
2. Assuming all of your IF stages had the same gain as the value in step 20, what was the overall gain of the section?
3. Using the results from steps 17 and 24 in equation 4.3, compute the Q.
4. How well do your results for problems 1 and 3 agree? Are they supposed to agree?
5. Compute the reactance of each *bypass* capacitor in the IF section at the IF frequency.
6. Compare your results in problem 5 to the circuit diagram value of the resistors that the capacitors are bypassing.

experiment 5 AUTOMATIC GAIN CONTROL

INTRODUCTION. Almost all radios have some means of controlling the gain automatically. In this experiment we shall examine some of the more common AGC methods.

DISCUSSION. Intermediate frequency amplifier transistors have output characteristics more-or-less like the one shown in figure 5-1.

Let's suppose that we use this transistor in the IF circuit shown in figure 5-2. The center IF frequency will be

$$f_{IF} \approx \frac{1}{2\pi\sqrt{L_1C_1}} = \frac{159 \times 10^3}{\sqrt{0.124}} \approx 455 \text{ kHz}$$

And, let's suppose that the tuned circuit Q is about 30. This is a typical value for two-stage commercial AM broadcast radio IF.

The collector load at 455 kHz is then

$$R_L = QX_{L1} = 2\pi f_{IF} L_1 Q$$

$$R_L = 6.28 \times 455 \times 200 \times 10^{-3} \times 30$$

$$R_L \approx 17\text{k ohms}$$

The loadline for this value of load is shown on figure 5-1. If we determine the values of

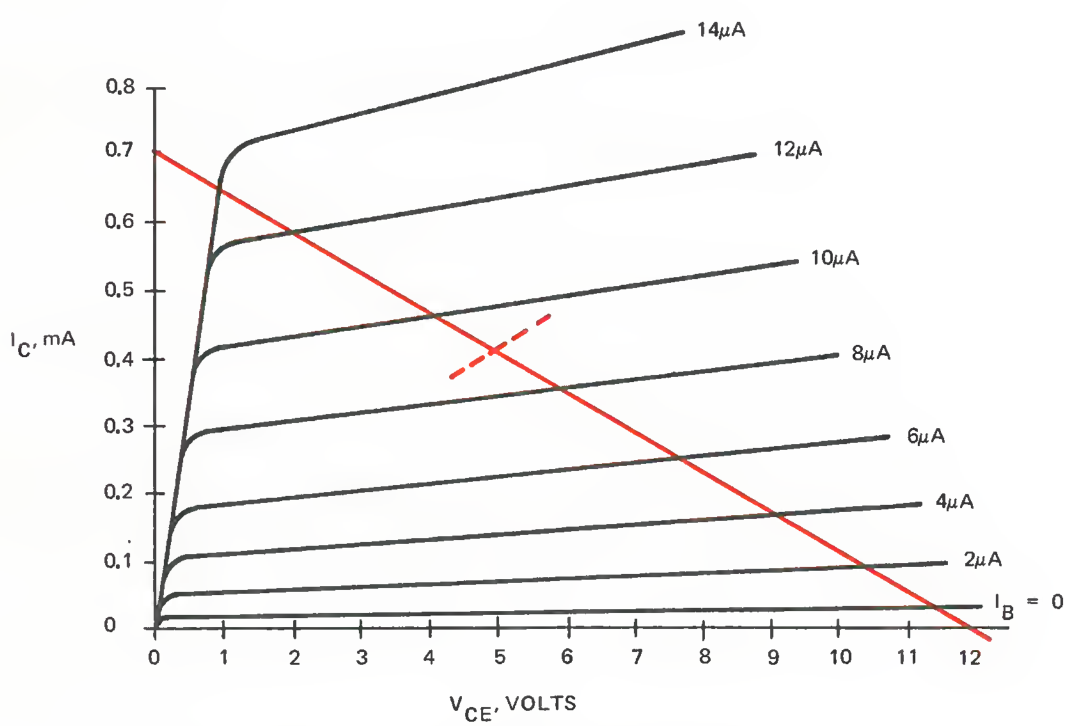


Fig. 5-1 A Transistor Output Characteristic

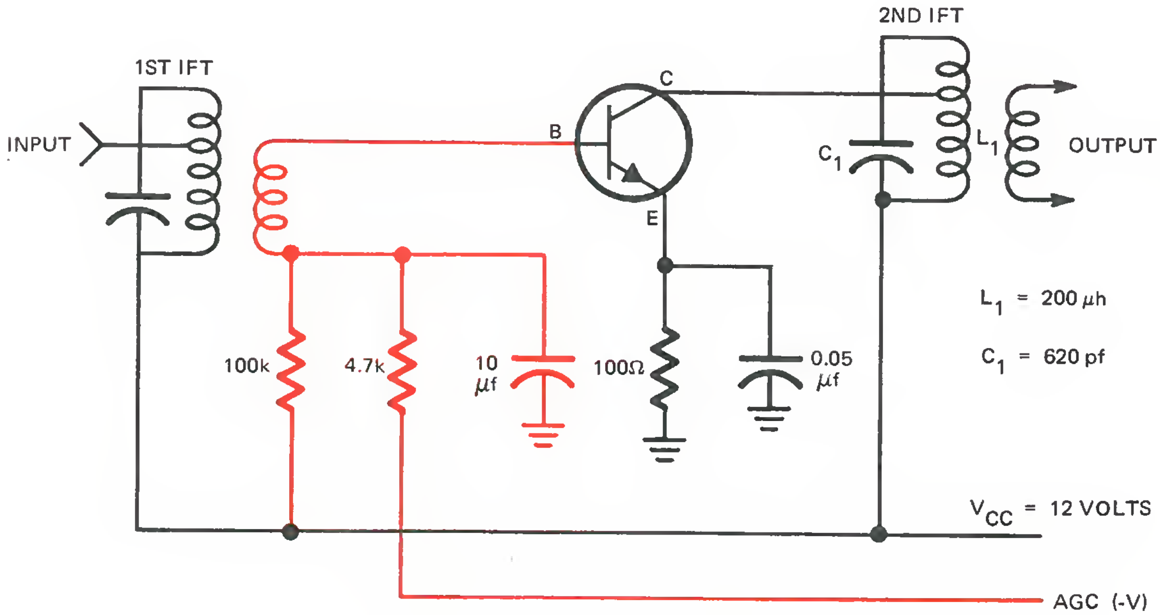


Fig. 5-2 An IF Amp with AGC

corresponding base and collector currents at points along this loadline we will get

$I_B(\mu A)$	0	2	4	6	8	10	12	14
$I_C(mA)$	0.03	0.09	0.17	0.25	0.36	0.47	0.58	0.66

Table 5-1 I_B vs I_C

Then if we go to the transistor input characteristic shown in figure 5-3, we can determine the values of base-emitter voltage that correspond to the base currents listed.

In this case the values will be

I_B (μA)	0	2	4	6	8	10	12	14
V_{BE} (volts)	0.4	0.55	0.61	0.66	0.69	0.71	0.72	0.73

Table 5-2 I_B vs V_{BE}

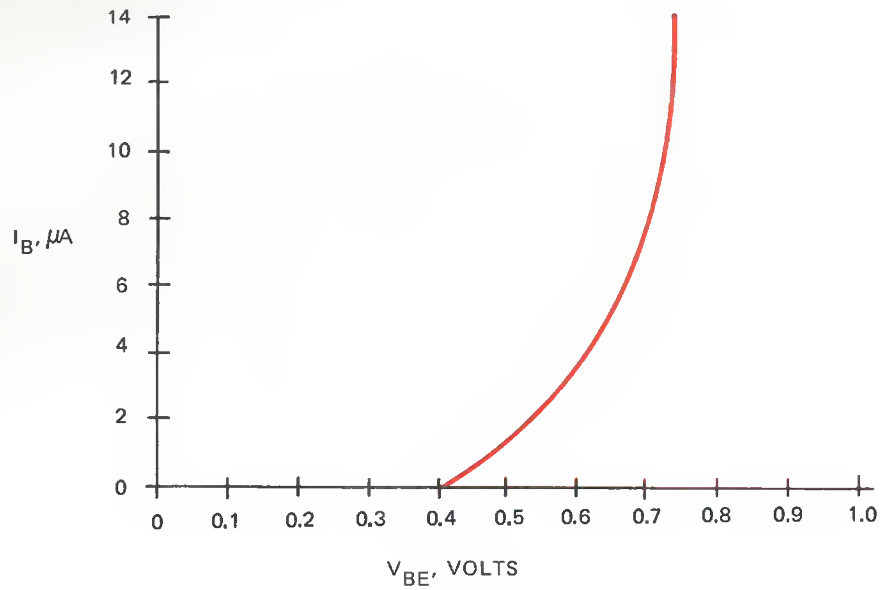


Fig. 5-3 The Transistor Input Characteristic

Now, from the circuit diagram (figure 5-2) we see that the base-to-ground voltage is given by

$$V_B = V_{BE} + V_E = V_{BE} + I_E R_E$$

And since I_B is very small $I_E \approx I_C$. So we can use

$$V_B = V_{BE} + I_C R_E$$

Calculating $I_C R_E$ using 100 ohms and I_C values from table 5-1, then adding the corresponding V_{BE} value from table 5-2 we have

I_B (μA)	0	2	4	6	8	10	12	14
V_B (volts)	0.40	0.56	0.63	0.69	0.73	0.76	0.78	0.80

Table 5-3 I_B vs V_B

Finally, since h_{fe} is approximated by

$$h_{fe} \approx \frac{\Delta I_C}{\Delta I_B}$$

we can calculate approximate h_{fe} values from table 5-1. These values are shown in table 5-4.

V_B (volts)	0.56	0.63	0.69	0.73	0.76	0.78
h_{fe}	35	40	47.5	55	55	47.5

Table 5-4 V_B vs h_{fe}

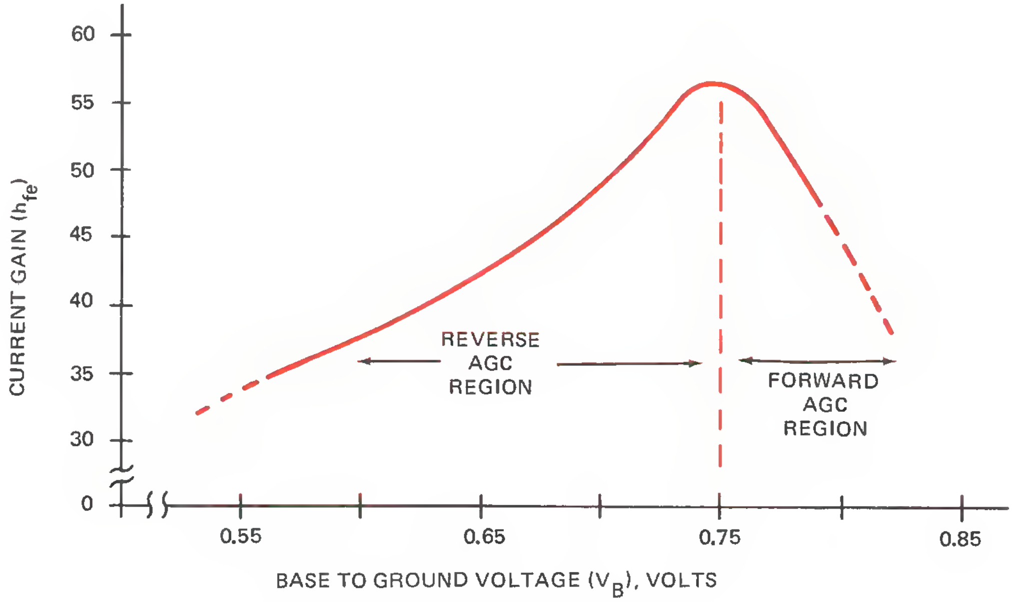


Fig. 5-4 Current Gain vs Base Voltage

If we plot these values against each other, the result will be similar to figure 5-4. Notice that h_{fe} is maximum at about 0.75 volts from base-to-ground. Both above and below this point h_{fe} decreases.

You will remember that the current gain of the stage is approximately equal to h_{fe}. Also the voltage gain is directly proportional to h_{fe}, so we see that both current gain and voltage gain *depend* on the value of the base voltage. And since the power gain is the product of the voltage and current gains, it too depends on the base voltage.

There is, of course, considerable variation in characteristics from one transistor to another. Therefore, the discussion presented here is intended only to illustrate the nature of parameter variation. *The actual numbers and curves used do not apply to any particular transistor.*

If we were going to build an AGC system using the data presented above, we would want the amplifier to have maximum gain with little or no signal input. We would, therefore, bias the stage to have a base voltage of approximately 0.74 volts with no input signal. This is done in figure 5-2 with a 100k and a 4.7k resistor between V_{CC} and the AGC point.

Now when the signal strength increases, we want the gain to decrease. From figure 5-4 we see that we have two choices:

1. We can use a negative AGC voltage to decrease the bias on the IF stage. This will reduce the gain.
2. We can use a positive AGC voltage to increase the bias on the IF stage. This will also reduce the stage gain.

The first of these methods is called *reverse AGC.* It is probably the most common type used in IF amplifiers. The second method is called *forward AGC* and it is only

rarely used in an IF stage. Forward AGC is frequently used in receiver RF amplifiers. Many receivers use both forward AGC in the RF amplifier and reverse AGC in the IF amplifier.

The AGC filter composed of the 4.7k resistor and the 10 μF capacitor in figure 5-2 must have a time constant that is long compared to the period of the modulation signal. In this particular case the time constant is

$$\tau = RC = 4.7 \times 10^3 \times 10 \times 10^{-6}$$
$$= 4.7 \times 10^{-2} \text{ sec} = 47 \text{ ms}$$

Let's suppose that the lowest frequency the modulation signal will have is 100 Hz. This is typical for an AM broadcast receiver. The period of this signai would be

$$t_m = \frac{1}{f_m} = \frac{1}{100} = 10 \text{ ms}$$

Notice that the AGC filter has a time constant that is about 5 times the longest modulation period.

The AGC time constant must not be so long that the receiver is sluggish in responding to changes in signal level. About 50 ms is typical for commercial AM radios.

Figure 5-5 shows a complete receiver in a schematic form. This receiver has reverse AGC of the usual type in the first IF amplifier stage. And it uses an AGC amplifier stage to produce forward AGC in the RF amplifier circuit. There are several things that we have not discussed that occur in this forward AGC circuit. Let's notice that in this RF amplifier the AGC voltage is applied to the collector circuit rather than to the base. If you will look back at figure 5-1, you will notice that there are three interdependent variables I_B, I_C, and V_{CE}. Previously we have controlled I_B by controlling V_B. When V_B is increased, I_B decreases. This effect produces forward AGC. We could just as well have controlled V_{CE} directly. To produce forward AGC we must cause V_{CE} to decrease when the signal level increases.

With this in mind let's go back to the receiver in figure 5-5. When the signal level increases, the IF AGC voltage goes more negative. In the NPN IF amplifier, this negative AGC reduces the bias producing reverse AGC.

The RF AGC voltage on the other hand goes positive as the signal increases. This positive AGC voltage increases the conduction of the AGC amplifier causing its collector voltage to decrease. Notice the AGC amplifier's collector voltage controls the collector voltage of the RF amplifier. So as the signal increases, the collector voltage of the RF amplifier decreases producing forward AGC.

We could have gotten forward AGC simply by connecting the positive AGC voltage directly to the RF amplifier base just like is done in the IF amplifier. This is indeed sometimes done.

Using an AGC amplifier let us include one feature that we haven't discussed at all yet. That is *delayed AGC*. In order to understand why delayed AGC is used, you need to observe that the signal level in the RF amplifier stage is normally very small. Under most conditions we don't want to reduce the gain of the RF amplifier. But sometimes an incoming signal will be so strong it will overdrive the first IF amplifier. Then we want AGC in the RF amplifier.

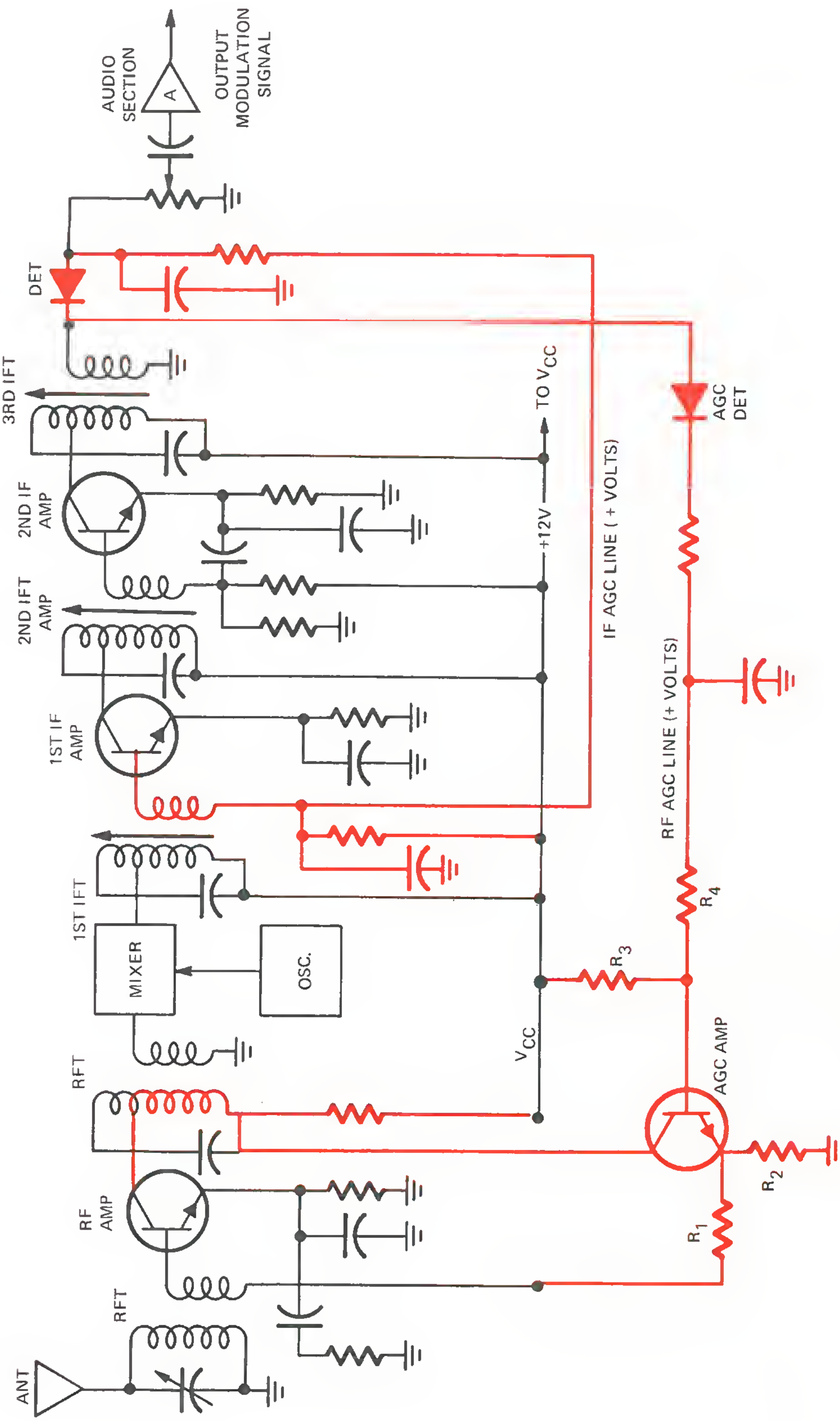


Fig. 5-5 A Receiver with Reverse IF AGC and Forward Delayed RF AGC

To get this effect, we hold the AGC amplifier cutoff until the AGC voltage reaches a particular value. Then we let the AGC amplifier come on and reduce the gain of the RF amplifier stage. In the AGC amplifier circuit, the emitter and base resistance (R_1, R_2, and R_3, R_4) are chosen so that the stage is cutoff until the AGC voltage gets up to the desired value.

The process of holding the RF AGC off until the signal reaches some preset value is called *delayed AGC.*

The chief difference between reverse and forward AGC lies in their respective signal handling abilities. Notice that in reverse AGC the controlled amplifier moves closer to the cutoff as the signal level increases. So, as the signal gets bigger, the amplifier gets less able to handle large signals. Forward AGC on the other hand moves the amplifier toward saturation. But since saturation in a transistor does not occur nearly as abruptly as cutoff does, forward AGC does not produce as much large signal distortion as reverse AGC.

MATERIALS

1 AM radio receiver
1 Circuit diagram for the radio
1 Oscilloscope
1 RF signal generator
1 VOM or FEM
1 Resistance substitution box (15Ω – 10 meg)
1 DC power supply (0 – 40V)
1 Sheet of linear graph paper

PROCEDURE

1. Turn on all of your equipment and allow it several minutes to warm up.
2. Loosely couple the RF signal generator to the receiver antenna. Set the generator for maximum output at about 1 MHz.
3. Connect the oscilloscope across the detector output.
4. Tune the receiver for maximum test signal at the oscilloscope.
5. Turn the receiver volume all the way down.
6. Disconnect the receiver AGC. This can be done by removing the AGC filter resistor.
7. Connect the DC power supply and resistance box to the first IF AGC point as shown in figure 5-6. Be very sure that you have the supply connected with the correct polarity. The figure shown is correct for a PNP amplifier. If your receiver uses NPN transistors you will need to use the opposite polarity.

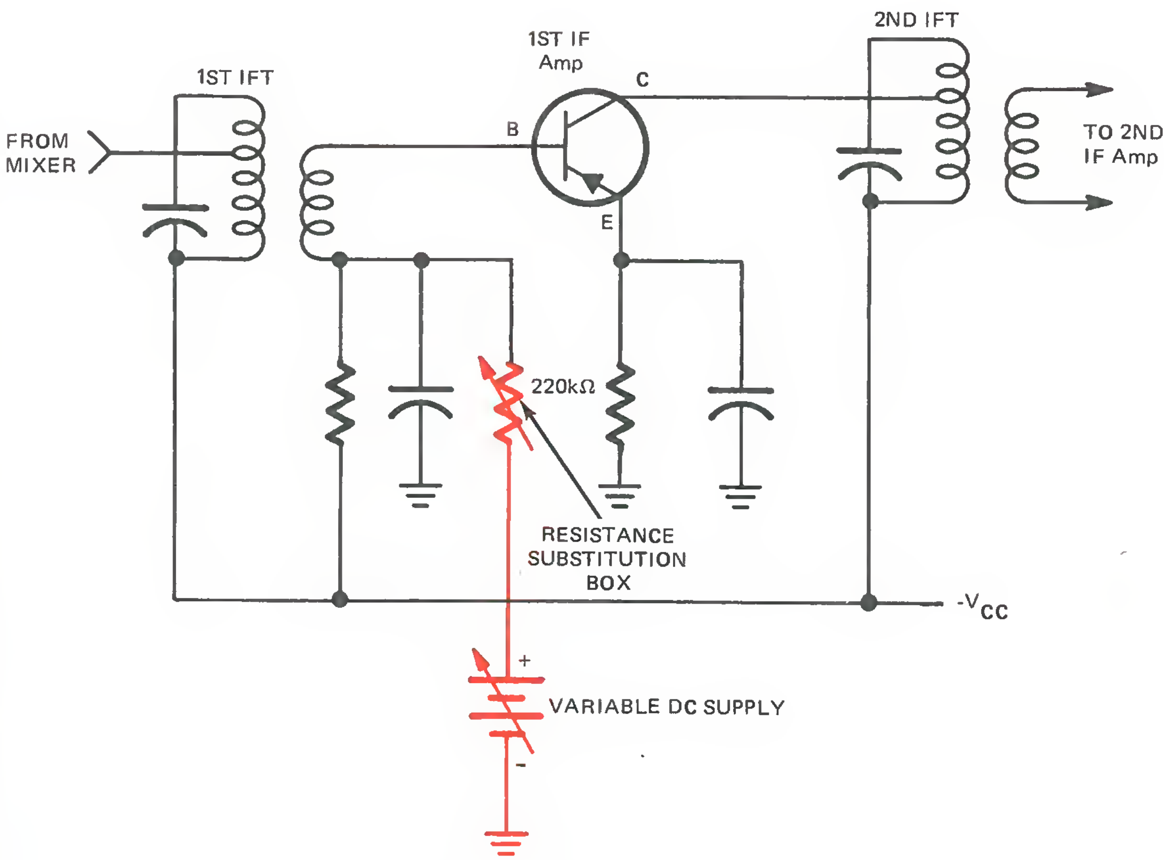


Fig. 5-6 Connecting the AGC Supply

8. Set the power supply for zero volts.
9. Measure and record the peak-to-peak output voltage and the DC base voltage.
10. Increase the *power supply voltage* in increments of one volt and repeat step 9 for each value until the peak-to-peak signal level is zero.
11. Return the power supply voltage to zero volts.
12. Reverse the polarity of the power supply. Then repeat steps 9, 10, and 11.
13. On a sheet of linear graph paper plot the peak-to-peak output voltage versus the base voltage.
14. Remove the resistance box and reconnect the receiver AGC.

ANALYSIS GUIDE. In the analysis of your data you should consider the following points: did your results agree with the material presented in the discussion; to what extent does your curve resemble figure 5-4? Based on your results which is more linear, reverse or forward AGC? Which is more gradual? Which cuts off most abruptly?

First Circuit

Base Voltage	Peak-Peak Output Voltage

Second Circuit

Base Voltage	Peak-Peak Output Voltage

Fig. 5-7 The Data Tables

PROBLEMS

1. Using the values from the circuit diagram of your receiver, what is the approximate value of the AGC time constant?
2. What would you expect the lowest modulation frequency to be that your set would reproduce? (Use your answer in problem 1 to estimate this answer).
3. Does your receiver use reverse or forward AGC? Explain why you think your answer is correct.
4. What would happen to the AGC if the detector diode were put in backward?
5. For each of your data points, compute the gain loss in decibels.

$$\text{Loss} = 20 \log \frac{E_0 \text{ max}}{E_0}$$

6. Plot the loss in db versus the base voltage. How does this compare with your original plot?

experiment 6 OSCILLATORS

INTRODUCTION. Superheterodyne receivers use an oscillator circuit to convert the incoming RF signal to the IF signal. In this experiment we shall examine some basic oscillator principles and circuits.

DISCUSSION. There are many different types of oscillator circuits. However they all depend on one concept of operation—positive feedback. Figure 6-1 shows a generalized feedback circuit. The overall gain of such a system is

$$\boxed{\frac{e_0}{e_i} = \frac{A}{1 - A\beta}} \qquad (6.1)$$

where A is the open loop gain of the amplifier and β is the feedback factor.

In this equation notice that the overall gain becomes undefined when

$$A\beta = +1$$

When this happens the system oscillates. To produce a stable oscillator we usually try to arrange the circuit so that $A\beta = 1$ at only one frequency. To do this we normally use a resonant circuit as the feedback network. Figure 6-2 shows one of the most commonly encountered oscillator circuits. In this circuit the transistor provides the gain. It is biased by the base and emitter resistors R_1, R_2, and R_3. C_2 is a bypass capacitor for the emitter resistor and C_3 couples the signal out of the oscillator circuit. The transformer does two things: it provides the feedback from collector to base, and it tunes the oscillator to the desired frequency.

The resonant frequency of the tuned circuit *(tank)* is approximately

$$f_0 = \frac{1}{2\pi\sqrt{L_1C_1}}$$

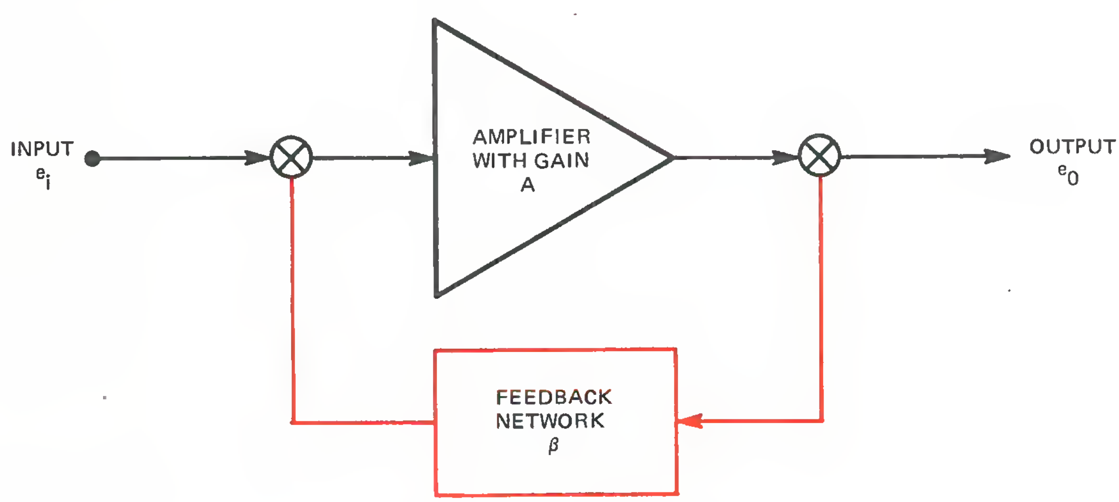


Fig. 6-1 A Feedback System

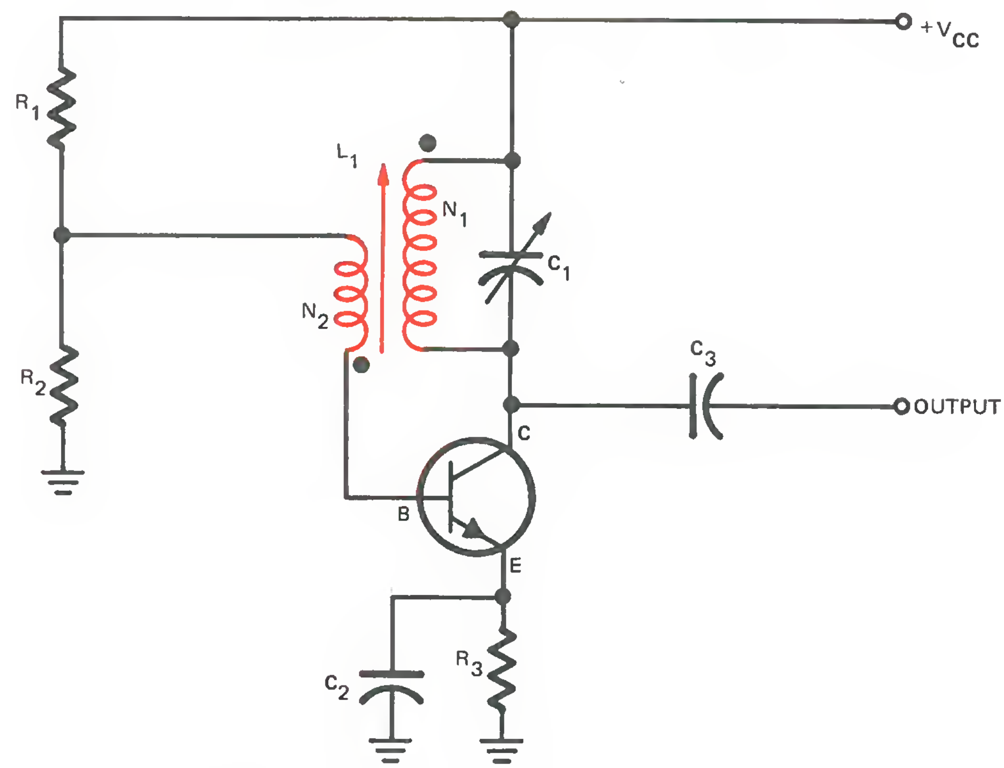


Fig. 6-2 A Transformer-Coupled Oscillator

and the feedback factor is approximately

$$\beta = \frac{N_2}{N_1}$$

Then since the current circulating in the tank is

$$i_t \approx Qi_C$$

where Q is the Q of the tank. Also because of transformer action

$$i_t = \frac{N_1}{N_2} i_B$$

Therefore

$$Qi_C \approx \frac{N_1}{N_2} i_B$$

or

$$\frac{i_C}{i_B} \approx \frac{N_1}{QN_2}$$

But since i_C/i_B is h_{fe} we have

$$\boxed{h_{fe} \approx \frac{N_1}{QN_2}} \qquad (6.2)$$

as the condition which will produce oscillation. Actually, in order to insure stable oscillation we usually choose a transistor which has a *minimum* h_{fe} greater than N_1/QN_2.

If we have an oscillator transistor and we wish to wind a coil for it, we can use this equation as a first approximation of the coil windings we need.

Sometimes the oscillator circuit given in figure 6-2 is modified by making the base side of the transformer tunable rather than the

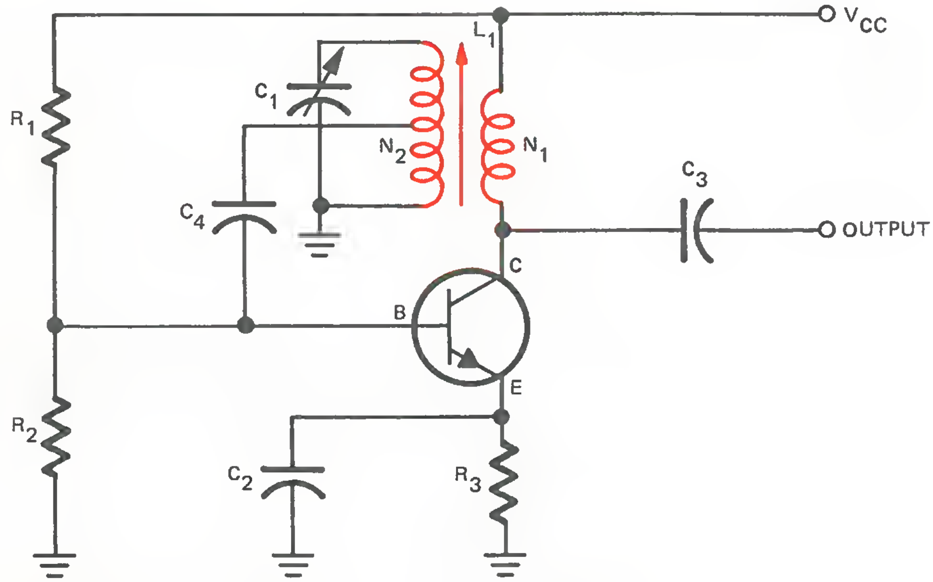


Fig. 6-3 A Tapped Tank Transformer-Coupled Oscillator

collector side. Figure 6-3 shows an oscillator circuit of this type. In this circuit the tuned circuit inductance is tapped to match impedances between the collector and the base.

The capacitor C_4 couples the signal from the tank into the base circuit. In this case we are using voltage feedback instead of current feedback as was used in figure 6-2.

Transformer coupling in an oscillator offers the advantage that the feedback can always be made regenerative simply by selecting the appropriate winding polarity.

Many oscillator circuits do not use this type of feedback. Figure 6-4 shows a circuit which employs a tapped coil for feedback.

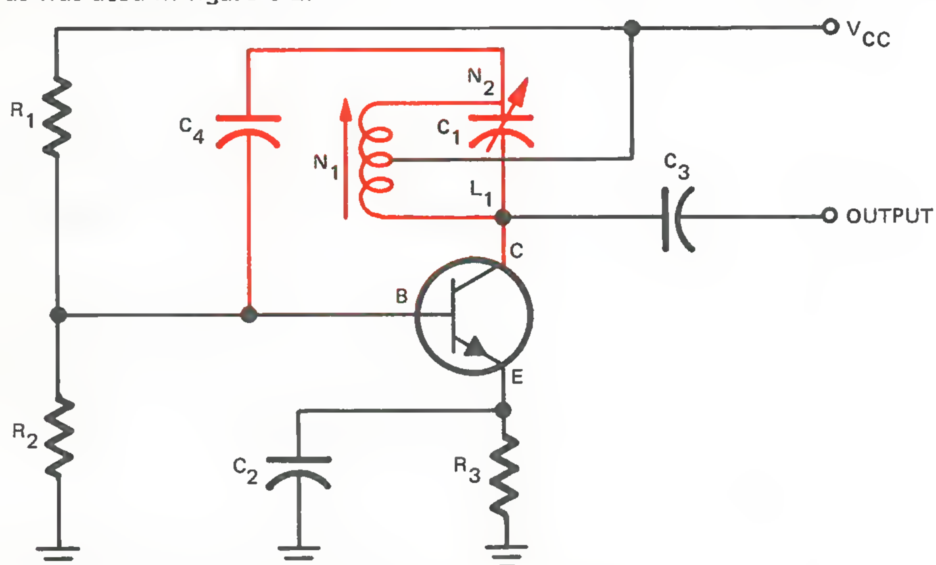


Fig. 6-4 A Hartley Oscillator Circuit

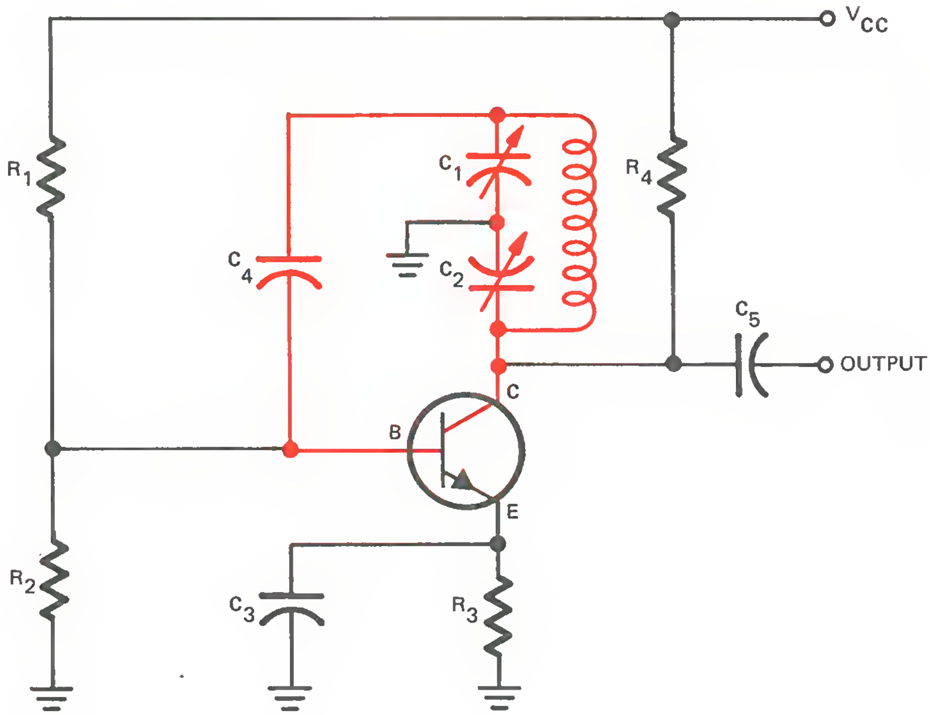


Fig. 6-5 A Colpitts Oscillator Circuit

Such a circuit is called a *Hartley* oscillator. This type of circuit is more frequently used in transmitters than in receivers. The tapped coil acts as an autotransformer with the feedback being the voltage between the tap and the upper end of the coil.

Another similar type of oscillator circuit is the *Colpitts* oscillator shown in figure 6-5.

The principal differences between the Colpitts and Hartley circuits are the methods of tapping off the feedback and the collector supply connections. The Colpitts circuit uses a capacitive voltage divider whereas the Hartley oscillator uses a tapped coil. The Hartley circuit is connected to the collector supply at the coil tap while the Colpitts oscillator is supplied through a collector resistor R_4. Since R_4 is shunted effectively across part of the tank it tends to lower the circuit Q. To reduce this loading effect an RF choke coil may be used instead of R_4.

Quartz crystals are frequently used as oscillator tuned circuits at the higher radio frequencies. Figure 6-6 shows a typical crystal oscillator circuit. The equivalent circuit of the crystal itself is also shown. Crystals have the advantage of very high Q, that is, R in the equivalent circuit is extremely small. This type of circuit where the crystal is used as the feedback circuit is called a *Pierce* oscillator.

As is the case with LC oscillators there are several different crystal circuits that are used. They all use the crystal as the device for determining the oscillator frequency and its value can be held within extremely close tolerances.

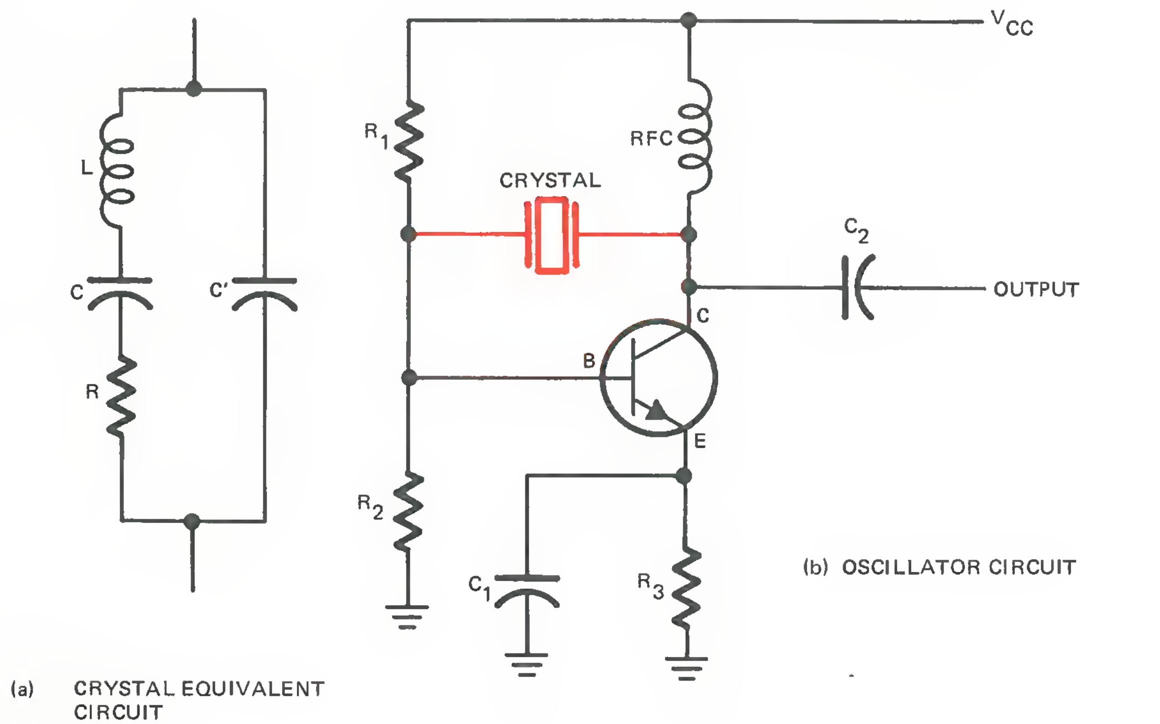


Fig. 6-6 A Typical Pierce Oscillator

When an oscillator is used in a superheterodyne radio, it must be capable of operating over a range appropriate for the receiver. For example, if an AM broadcast radio is to operate from 530 kHz to 1610 kHz and the IF frequency is 455 kHz, then the oscillator must operate from 985 kHz (530 + 455) up to 2065 kHz (1610 + 455).

Actually most receivers have a slightly broader oscillator range to allow for a little oscillator frequency variation.

In adjusting the oscillator in a receiver there are two adjustments to be made. They are the tank capacitor *trimmer* and the coil inductance. These two adjustments interact with one another to a considerable extent. However, most receivers will operate somewhat with the oscillator coil slug set near the middle of its range of adjustment. This fact provides us with a starting point. With the coil slug near its mid position we set the tuning dial to some relatively high frequency – say 1400 kHz for a broadcast radio. Then adjust the trimmer on the side of the tuning capacitor for the correct output frequency – 1855 kHz in this case. When this is done turn the dial down to a relatively low frequency – 600 kHz will do fine. *Now adjust the slug* in the oscillator coil for the correct output frequency (1055 kHz now). After the coil is adjusted go back to a dial setting of 1400 kHz and again adjust the *trimmer* for an output of 1855 kHz. Then do the slug adjustment again at a dial setting of 600 kHz. Repeat this process of adjusting the trimmer at a high frequency and the coil at a low frequency until no further changes in the settings are needed. The oscillator is now working *approximately correctly.*

To improve the tuning first align the IF stages correctly. Then apply a modulated 1400-kHz signal to the receiver input. Set the

receiver dial to 1400 kHz and adjust the oscillator trimmer for maximum modulation output. Then apply a 600-kHz modulated input, set the receiver to 600 kHz and adjust the coil for maximum modulation output. Repeat this process as before until no further improvement in output is experienced. The oscillator is now correctly adjusted. It is always a good idea to check the receiver at several other frequencies, say 500 kHz, 800 kHz, 1200 kHz, and 1600 kHz to insure that there is no failure of oscillation as it is tuned.

MATERIALS

- 1 AM radio receiver
- 1 Schematic diagram for the receiver
- 1 RF signal generator
- 1 Oscilloscope
- 1 Nonmetallic hex type tuning tool
- 1 Nonmetallic screw driver type tuning tool

PROCEDURE

1. Turn on all your equipment and allow it several minutes to warm up.
2. With your tuning tool set the slug in the oscillator coil to about the middle of its adjustment range. Note – This coil is *not* ruggedly constructed. Take reasonable care in working with it.
3. Connect the oscilloscope across the oscillator tuning capacitor. *Use either* a low capacitance probe or a properly compensated x10 probe.
4. Set the receiver dial to 1400 kHz and adjust the trimmer screw on the side of the oscillator tuning capacitor until the output frequency is as near 1855 kHz as you can get it.
5. Set the receiver dial to 600 kHz and carefully adjust the oscillator coil slug until the oscillator frequency is as near 1055 kHz as you can get it.
6. Repeat steps 4 then 5 several times until no further adjustment of the trimmer or coil are needed.
7. Move the oscilloscope to the output of the detector stage. Set the receiver dial to its maximum frequency setting.
8. Loosely couple a 455-kHz modulated signal from the RF generator in to the antenna of the receiver. You should be able to hear the test tone. If so, turn the volume down and use the scope as an output indicator.
9. Align the IF stages. Always keep the generator level low so that you will not cause the AGC to reduce the IF gain. Work from the detector toward the front end of the receiver and adjust each IFT for maximum modulation signal. Go over the IF adjustments several times to insure sharp alignment.
10. Set the signal generator frequency to 1400 kHz. Set the receiver dial to 1400 kHz and the oscillator trimmer for maximum modulation output.
11. Set the signal generator frequency to 600 kHz. Set the receiver dial to 600 kHz and adjust the oscillator coil slug for maximum modulation output.

12. Repeat step 10 then 11 several times until no further improvement in modulation output is observed.
13. Move the oscilloscope back to the oscillator as it was in step 3.
14. Set the signal generator to 540 kHz. Adjust the receiver dial until you hear the best test tone at the speaker. Keep the volume level as low as practical. Record the generator frequency (f_{RF}), the oscillator frequency (f_0), and the receiver dial setting (f_d) in the data table.
15. Repeat step 14 for generator frequencies of 600, 700, 800, 1000, 1300, and 1600 kHz.
16. For each set of data compute $f_{IF} = f_0 - f_{RF}$. Enter your results in the data table.
17. Set the receiver dial to its minimum setting. Then adjust the RF generator around 550 kHz until you get the best test tone. Record the generator frequency (f'_{RF}) and the oscillator frequency (f'_0).
18. Now increase the RF generator frequency to around 1450 kHz until you again hear the best test tone. Record the generator frequency (f''_{RF}).

f_{RF}	f_0	f_d	f_{IF}

f'_{RF}	f'_0	f''_{RF}

Fig. 6-7 The Data Tables

ANALYSIS GUIDE. In analyzing your results from this experiment you should consider several points. Among them are: How well do the dial readings agree with the incoming signal frequency? Did the IF frequency stay constant across the tuning range of the receiver? Why did you get two different incoming frequencies for only one oscillator setting in steps 17 and 18?

PROBLEMS

1. Explain in your own words how the oscillators in figures 6-2 and 6-3 work.
2. Repeat problem 1 for figures 6-4 and 6-5.
3. The effective inductance of the tank circuit in figure 6-2 is 30 μH. What range of capacitance must the capacitor have if the oscillator is to be used in an AM broadcast radio?
4. Explain how you would quickly check an oscillator in a malfunctioning receiver.
5. What is the main advantage of using a crystal oscillator?
6. A certain FM receiver has an IF frequency of 10.4 MHz. It is tuned to 96.4 MHz on its dial. A strange signal keeps interfering with the reception. What are two frequencies on which the interference could be operating? Explain your answer.
7. Explain how the oscillator in figure 6-6 operates.

experiment 7 HETERODYNING

INTRODUCTION. Heterodyning is one of the basic principles upon which modern communications often depends. In this experiment we shall examine the principle of heterodyning and some of the heterodyning circuits used in radios.

DISCUSSION. Let's examine the operation of the circuit diagram shown in figure 7-1. The inputs to this circuit are two radio frequency signals. They are both sinusoidal and one we will call the RF signal input, e_s. This signal input we will assume to be of the form

$$e_s = E_s \cos \omega_s t$$

The other signal we will call the *oscillator* input, e_o, and assume that it has the form

$$e_o = E_o \cos \omega_o t$$

The total input voltage is the sum of these two input components

$$e_T = e_s + e_o = E_s \cos \omega_s t + E_o \cos \omega_o t \qquad (7.1)$$

This voltage acts to cause a current, i, to flow in the circuit.

The circuit current must flow through the diode which is a *square law device.* We can, therefore, take the circuit current to be

$$\boxed{i = A_2 e_T^2} \qquad (7.2)$$

Substituting the equation for e_T into this expression gives us

$$i = A_2(E_s \cos \omega_s t + E_o \cos \omega_o t)^2$$

or

$$i = A_2 E_s^2 \cos^2 \omega_s t + A_2 E_s E_o \cos \omega_s t \cos \omega_o t + A_2 E_o^2 \cos^2 \omega_o t$$

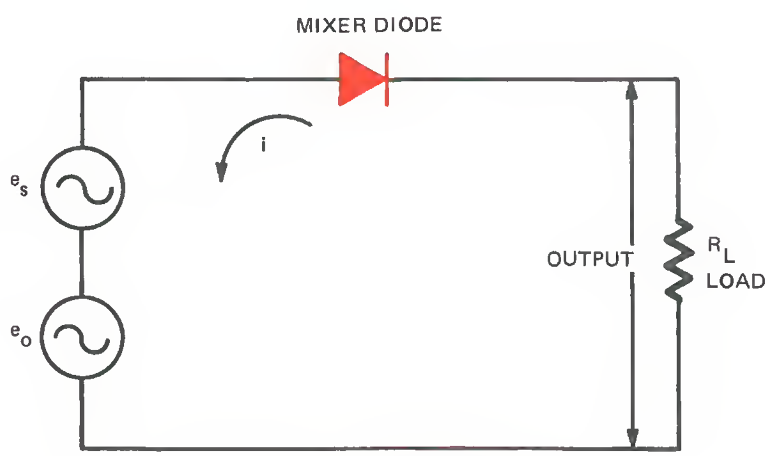


Fig. 7-1 A Simple Heterodyning Network

Then, if we apply the trigonometric identities

$$\cos^2 \omega t = \frac{1}{2} + \frac{1}{2} \cos 2\omega t$$

and

$$\cos \omega_s t \cos \omega_o t = \frac{1}{2} [\cos (\omega_s + \omega_o)t + \cos (\omega_s - \omega_o)t]$$

we will have

$$\begin{aligned}
i = {} & \tfrac{1}{2} A_2(E_s{}^2 + E_o{}^2) && \text{DC term} \\
& + \tfrac{1}{2} A_2 E_s{}^2 \cos 2\omega_s t && \\
& + \tfrac{1}{2} A_2 E_o{}^2 \cos 2\omega_o t && \text{high frequency terms} \\
& + \tfrac{1}{2} A_2 E_s E_o \cos (\omega_o + \omega_s)t && \text{sum term} \\
& + A_2 E_s E_o \cos (\omega_o - \omega_s)t && \text{difference term}
\end{aligned} \tag{7.3}$$

as the current through the load. Let's examine each of the terms one at a time.

The first term, $1/2\ A_2(E_s{}^2 + E_o{}^2)$, is a direct current term and is not normally used in the heterodyne process. It could, however, be used for AGC in the RF stage of a receiver. The next three terms are high frequency terms. The first one of these, $1/2\ A_2E_s{}^2 \cos 2\omega_s t$, is the second harmonic of the *signal* input. The next term, $1/2\ A_2E_o{}^2 \cos 2\omega_o t$, is the second harmonic of the *oscillator* input. The third high frequency term, $1/2\ A_2E_sE_o \cos (\omega_o + \omega_s)t$, is at a frequency which is equal to the sum of the *signal* input and *oscillator* input frequencies. Let's suppose that our two signals are the input and oscillator signals in a typical AM broadcast radio. Let's say that e_s is at a frequency of 1000 kHz ($\omega_s = 2\pi f_s = 6.28 \times 10^6$) and e_o is at 1455 kHz ($\omega_o = 2\pi f_o = 9.14 \times 10^6$). Then the three high frequency terms will be at frequencies of 2000, 2910, and 2455 kHz respectively.

The last term in equation 7.3 is at a frequency which is the *difference* between f_s and f_o. In our example this would be 455 kHz. It is this difference in frequency that we use for the IF signal and it is produced by *heterodyning.*

Let's observe at this point that each of the terms are multiplied by A_2, the diode circuits square law coefficient. Since A_2 typically runs in order of 10^{-6} all of the output terms will be very small in amplitude.

In an AM radio the incoming signal will normally be a carrier and two sidebands of the form

$$e_s = E_c \cos \omega_c t + \frac{1}{2} E_m \cos (\omega_c + \omega_m)t + \frac{1}{2} E_m \cos (\omega_c - \omega_m)t$$

When these three terms are heterodyned with the oscillator signal the *difference* terms become

$$\begin{aligned}
i = {} & A_2E_cE_o \cos (\omega_o - \omega_c)t \\
& + \tfrac{1}{2} A_2E_mE_o \cos (\omega_o + \omega_m)t \\
& + \tfrac{1}{2} A_2E_mE_o \cos (\omega_o - \omega_m)t
\end{aligned}$$

This is the IF signal normally fed to the IF section of the receiver. The first term is the IF carrier at the IF frequency (455 kHz in the example above). The second term is the upper IF sideband. Suppose the modulation signal was at 3 kHz, then this sideband would

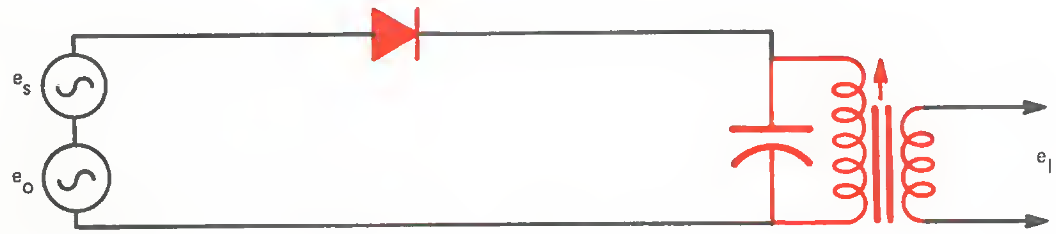


Fig. 7-2 Equivalent Circuit of a Mixer Stage

be at 458 kHz. The third term is the lower IF sideband at 452 kHz. You should take particular note of the fact that the sidebands of the IF signal are different from the IF carrier by the same amount as the sidebands of the incoming signal were different from the incoming carrier. In other words, the IF signal has the same bandwidth as did the incoming RF signal.

In most radios the load of the mixer diode is the first IF transformer. Figure 7-2 shows a more typical equivalent circuit for a mixer stage. The first IF transformer acts as a resonant filter which couples only the difference (IF) signal into the IF section. The high-frequency terms are bypassed through the capacitor and are not coupled into the IF section.

Mixer circuits of this type are used in practical receivers. Figure 7-3 shows a typical receiver front end of this type. The RF input signal is coupled into the mixer, D_1, from the

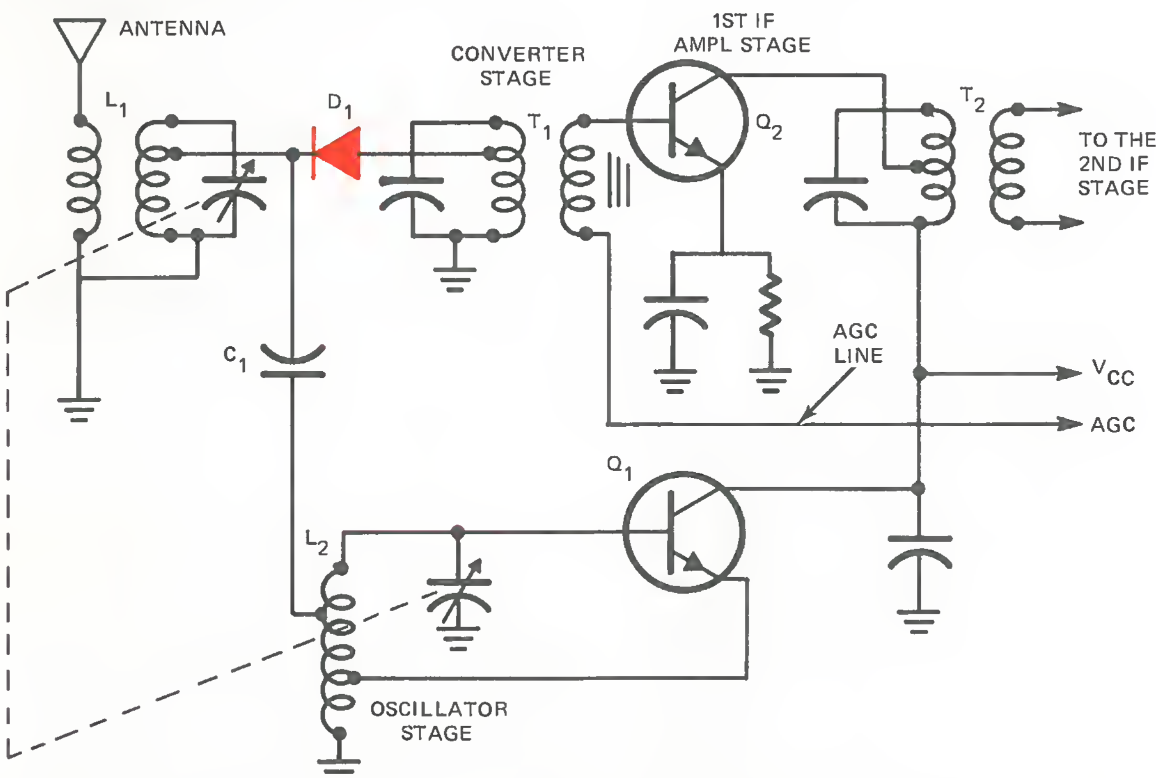


Fig. 7-3 A Receiver Front End

tapped antenna transformer. The input circuit is tuned to the frequency of the incoming signal.

The oscillator Q_1 is tuned to a frequency which differs from that of the incoming signal by an amount equal to the IF frequency. The oscillator output signal is coupled into the mixer through capacitor C_1. The first IF transformer, T_1, is tapped to help match impedances between the mixer output and the first IF amplifier's input.

Many receivers use a transistor mixer instead of a diode. Figure 7-4 shows such a circuit. The mixing operation is substantially the same using the base-emitter junction of Q_3 as the mixing diode. However, since the base current (mixer current) is amplified through the transistor, the square law coeffi-

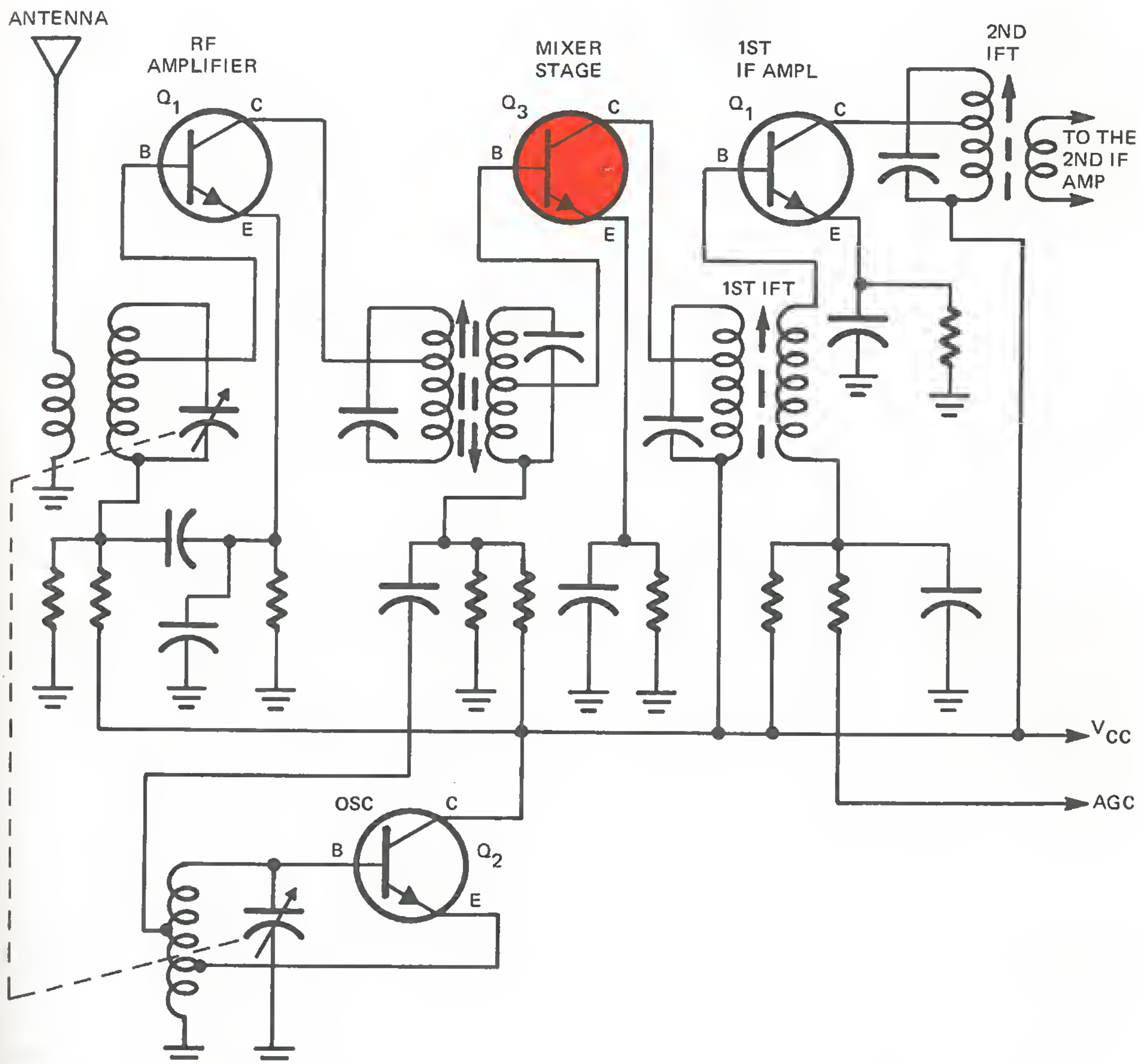


Fig. 7-4 A Transistor Mixer Stage

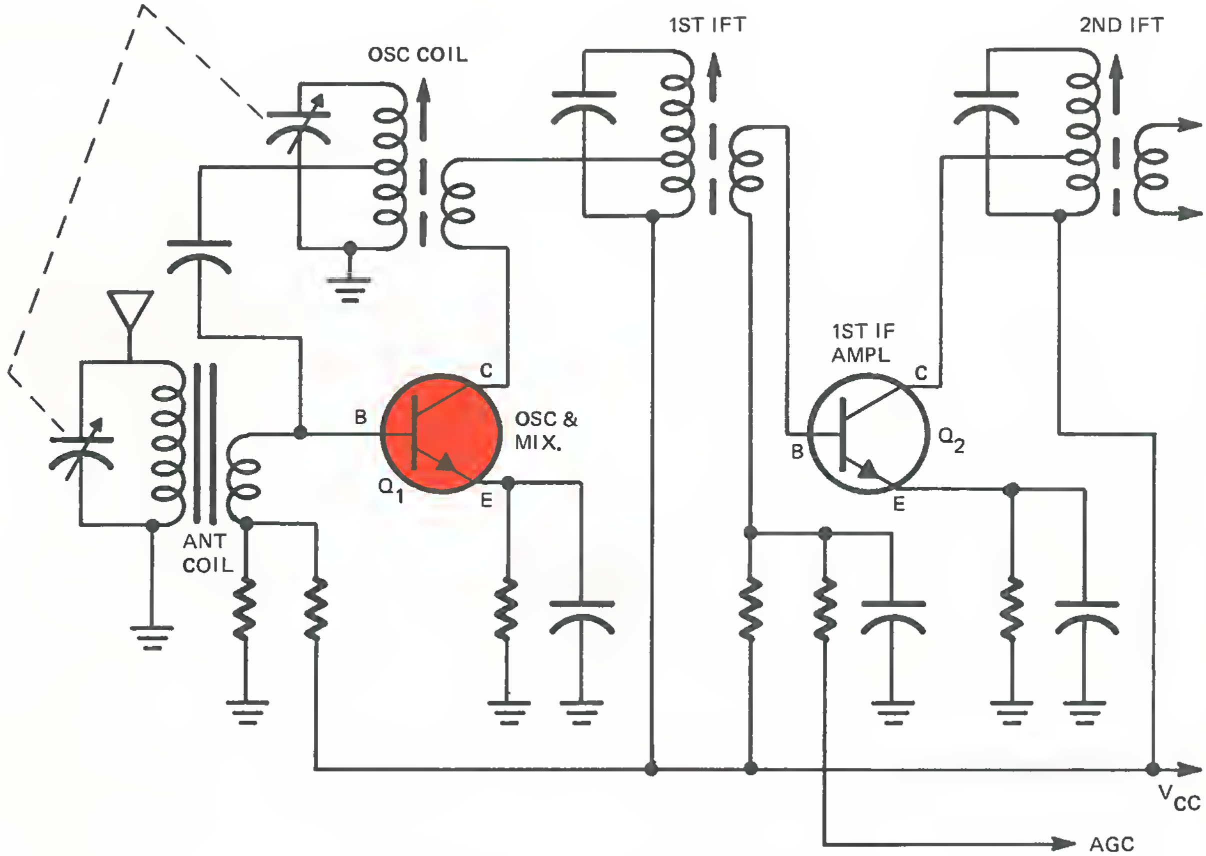


Fig. 7-5 An Autodyne Front End

cient A_2 is much larger than for a simple diode. Even so, in this heterodyning process, we get less IF energy output than that of the input RF signal.

Some receivers employ a circuit of the type shown in figure 7-5. This circuit uses the same transistor Q_1 for the oscillator and the mixer. The oscillator is simply a transformer-coupled circuit. However, at the base both the oscillator feedback signal and the incoming RF signal are injected. Mixing occurs in the base junction and the output is directly coupled to the first IF transformer. A circuit of this type, where a single stage is used as the oscillator and the mixer, is often called an *autodyne circuit.*

Perhaps the most serious problem that arises out of the use of heterodyning in a radio receiver is *image frequency interference.* To understand the nature of this type of interference, let's suppose that the oscillator in a receiver is operating at 1050 kHz. Now if there is an input signal at 595 kHz, the two will heterodyne producing a 455 kHz IF signal (455 = 1050 – 595), and we will hear the 595 kHz station. On the other hand, if there is an incoming signal at 1505 kHz, the oscillator will heterodyne with it and produce a 455 kHz IF signal (455 = 1505 – 1050), and we will hear the 1050 kHz station. Now, if both of the signals given above (595 kHz *and* 1505 kHz are applied to the mixer, then the oscillator will heterodyne with *both* of them and we will hear both at the same time. Or in

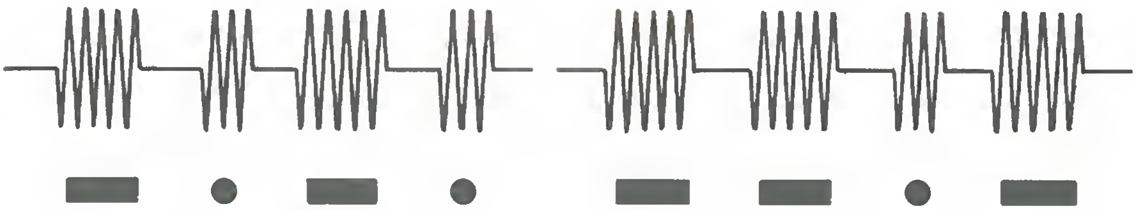


Fig. 7-6 A CW Telegraphic Signal

other words, the two signals would interfere with each other. It is common practice to tune the oscillator frequency *above* the desired incoming frequency. This being the case, we would call the 595 kHz signal the desired one and the 1505 kHz signal the interfering *image* signal.

The ability of a receiver to reject undesirable image signals is an important design consideration. Image rejection is achieved mainly by the use of very high Q tuning circuits in the RF section ahead of the mixer stage. For instance, if the antenna of the receiver used in the image interference example had been very sharply tuned to 595 kHz, then the undesired image could not have reached the mixer. The use of a sharply-tuned RF amplifier also enhances image rejection. Image rejection can also be improved by using IF amplifiers whose frequency response curve has very steep sides. If this is the case, then the image IF must be exactly the correct frequency or it won't get through the IF section even if it does slip through the RF tuning

Many AM radios do not have an amplifier stage. When this is the case, the mixer input must be tuned to the correct frequency. This is usually done by loosely coupling a 1600 kHz modulated signal into the antenna. Then the receiver dial is set to 1600 kHz and the antenna trimmer is adjusted for maximum receiver output. The antenna trimmer is an adjustment screw located on the side of the antenna section of the tuning capacitor.

Heterodyning is used in another way in receivers designed to receive telegraphy messages. Many telegraphy signals are sent with *continuous wave (CW)* radiation. In this type of service there is no modulation in the usual sense of the word. The carrier energy is simply switched on and off by the telegraphic signal. Figure 7-6 shows a graphic representation of such a CW signal.

Let's suppose that this signal has a frequency of 7200 kHz and that the receiver's oscillator is operating at 7655 kHz. The IF signal would then be at 455 kHz and would look like figure 7-6. The trouble is that changes in signal amplitude are at such a slow rate that we can't hear it. To make the CW signal audible we can heterodyne it with another signal at the detector. Figure 7-7 shows this type of heterodyning. The oscillator in this case is called a *Beat Frequency Oscillator* or *BFO.* It is normally designed to operate at the IF frequency and is tunable for about 1 kHz or 2 kHz above and below the IF frequency.

When the CW is present, the BFO heterodynes with it to produce a 1- or 2-kHz audible note. When the CW signal is not present, there is no receiver output. Most receivers equipped with a BFO have provision for turning it off when it is not in use.

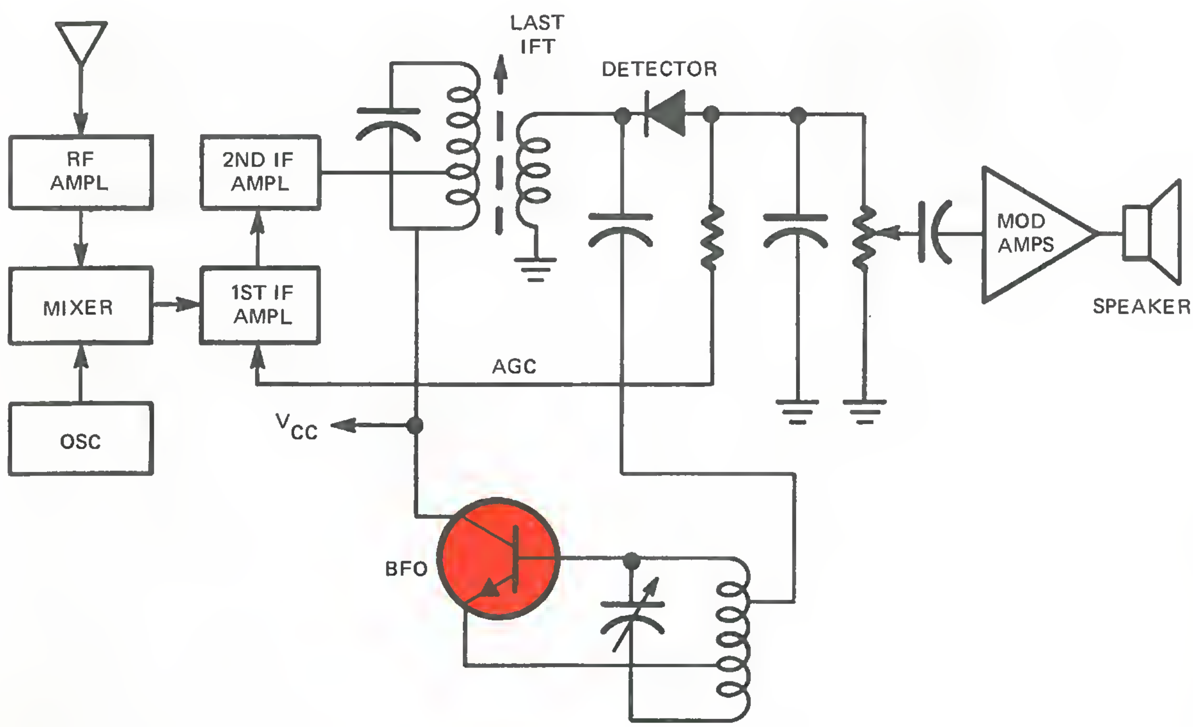


Fig. 7-7 A BFO Circuit

MATERIALS

1 AM radio receiver
1 Diagram for the receiver
1 RF signal generator
1 Oscilloscope
1 Nonmetallic hex type tuning tool
1 Nonmetallic screw driver type tuning tool

PROCEDURE

1. Examine your receiver circuit diagram and record the type of mixer circuit it has in the data table.
2. Turn on your equipment and allow it several minutes to warm up.
3. Loosely couple the RF signal generator to the receiver antenna.
4. Connect the oscilloscope across the volume control and set the volume to zero.
5. Align the receiver IF section.
6. Check the oscillator trimmer adjustment at 1400 kHz. Then check the oscillator coil adjustment at 600 kHz.
7. Set the signal generator and the receiver tuning dial to 1600 kHz.
8. Adjust the antenna trimmer on the side of the antenna tuning capacitor for maximum receiver output.

9. Connect the oscilloscope across the oscillator tuning capacitor. Use either a low capacity probe *or* a properly compensated x10 probe.

10. Set the signal generator to 550 kHz. Tune the receiver until you hear the signal generator test tone in the speaker. Record the signal generator frequency (f_{RF}) and the oscillator frequency (f_0) in the data table. In performing this step keep the volume level as low as practical.

11. Record the difference (f_{IF}) between the two values determined in step 10.

12. Repeat steps 10 and 11 for frequencies of 650, 750, 850, 950, 1050, 1150, 1250, 1350, 1450, and 1550 kHz.

13. Move the oscilloscope to the output of the first modulation amplifier. Set the volume control for a comfortable listening level.

14. Set the receiver dial to its highest setting. Then tune the RF generator to hear 455 kHz. Carefully tune the RF generator for maximum receiver output at the oscilloscope. Record the IF frequency (f'_{IF}).

15. Calculate the average IF frequency (f''_{IF}) from the data taken in steps 10, 11, and 12.

16. Connect the hot lead only (not the ground lead) from the RF signal generator to the output of the detector stage. This will allow a small amount of the generator signal to leak into the detector stage. Be careful not to break the diode.

17. Set the generator for maximum output at the IF frequency and *turn off the generator modulation.*

18. Now tune the receiver across its band and watch the oscilloscope. As you tune past each station you will see and hear an audio "whistle" from the set. Select one of the stations and set the receiver tuning to it.

19. Vary the RF signal generator from 5 kHz below the IF frequency to 5 kHz above the IF frequency. Record both the generator frequency f'_{RF} (about 5 points is enough, 2 above f'_{IF}, 2 below f'_{IF}, and 1 at f'_{IF}) and the audio tone frequency, f_B.

20. Calculate the difference (Δf) between f_{IF} and the RF generator frequencies taken in step 19.

ANALYSIS GUIDE. In the analysis of these data there are two main things you should consider. They are:

1. How well did f'_{IF} and f''_{IF} agree? Does the extent of their agreement verify the principles of heterodyning presented in the discussion? Why?

2. Do the values f_B and Δf seem to agree? Do these values tend to verify the principles of heterodyning as presented in the discussion? Why?

Type of Mixer Circuit = ____________________

f_{RF}	f_0	f_{IF}

f'_{RF}	f_B	Δf

f'_{IF} = ____________________

f''_{IF} = ____________________

Fig. 7-8 The Data Tables

PROBLEMS

1. If you had to call one of the data points in step 19 a *zero beat,* which one would it be? Why?

2. What is an image frequency? How could you produce image interference with your signal generator?

3. A certain receiver is picking up a 7.126 MHz CW signal. The receiver oscillator is operating at 7.578 MHz. What is the IF frequency?

4. The receiver in problem 3 has its BFO on and it is operating at 454 kHz. What would be the tone from the speaker.

5. The set in problem 3 is left on after the transmitting station on 7.126 MHz goes off the air. A weak 2-kHz tone is heard coming from the speaker. If this tone is a beat note, what are its two possible carrier frequencies?

6. How could you find out which of the two frequencies in problem 5 was the one being received? (Hint: What effect would changing the BFO frequency have in each case?)

experiment 8 RECEIVER TROUBLESHOOTING

INTRODUCTION. Radio receivers will occasionally fail to perform properly. In this experiment we will examine some of the techniques that can be used to locate the cause of a receiver failure.

DISCUSSION. A superheterodyne receiver has five different types of signals that it processes in one way or another. These signals are:

1. The modulated RF input signal.
2. The unmodulated oscillator signal.
3. The modulated IF signal.
4. The direct current AGC signal.
5. The modulation signal.

These five signals appear at different points in the receiver circuit. Figure 8-1 shows the block diagram of a receiver and the type of signal processed by each block.

We can use this knowledge of the type of signal present at each point in the receiver to help us localize a receiver problem. To illustrate how this can be done let's suppose we have a completely dead receiver. Our first step in troubleshooting would be to turn the receiver on and check the V_{CC} supply voltage. This can readily be done with a VOM. If the value of V_{CC} is normal then connect an *audio* test signal across the volume control. Turn the volume up and listen. If the test tone is heard in the speaker and has normal volume then the audio section is working.

On the other hand if a normal test tone can't be heard in the speaker then there is a failure of some kind in the audio section. In which case we proceed to troubleshoot the audio amplifiers.

Audio amplifier failures may be divided into a number of major categories. The following major divisions will include the vast majority of troubles.

1. An amplifier with a normal input signal may have no output at all.
2. An amplifier with a normal input may have seriously reduced output.
3. An amplifier with a normal input may produce a distorted output.
4. An amplifier may oscillate at some frequency.
5. An amplifier output may contain objectional amounts of noise or hum.

Each of these five types of trouble is discussed individually in the following paragraphs.

In troubleshooting an amplifier (or other electronic system), we normally make two basic assumptions upon which to proceed. The first is that the circuit was functional before the failure. (This is the same as assuming that there is no error in circuitry or design.) Secondly, we assume that there is only one problem to be located in the circuit. (This does not exclude the possibility of several different failures, but does let us deal with one problem at a time.) With these basic assumptions in mind, let us now proceed with each of the five types of troubles.

NO OUTPUT SIGNAL. If the input signal to an amplifier is normal and there is no output

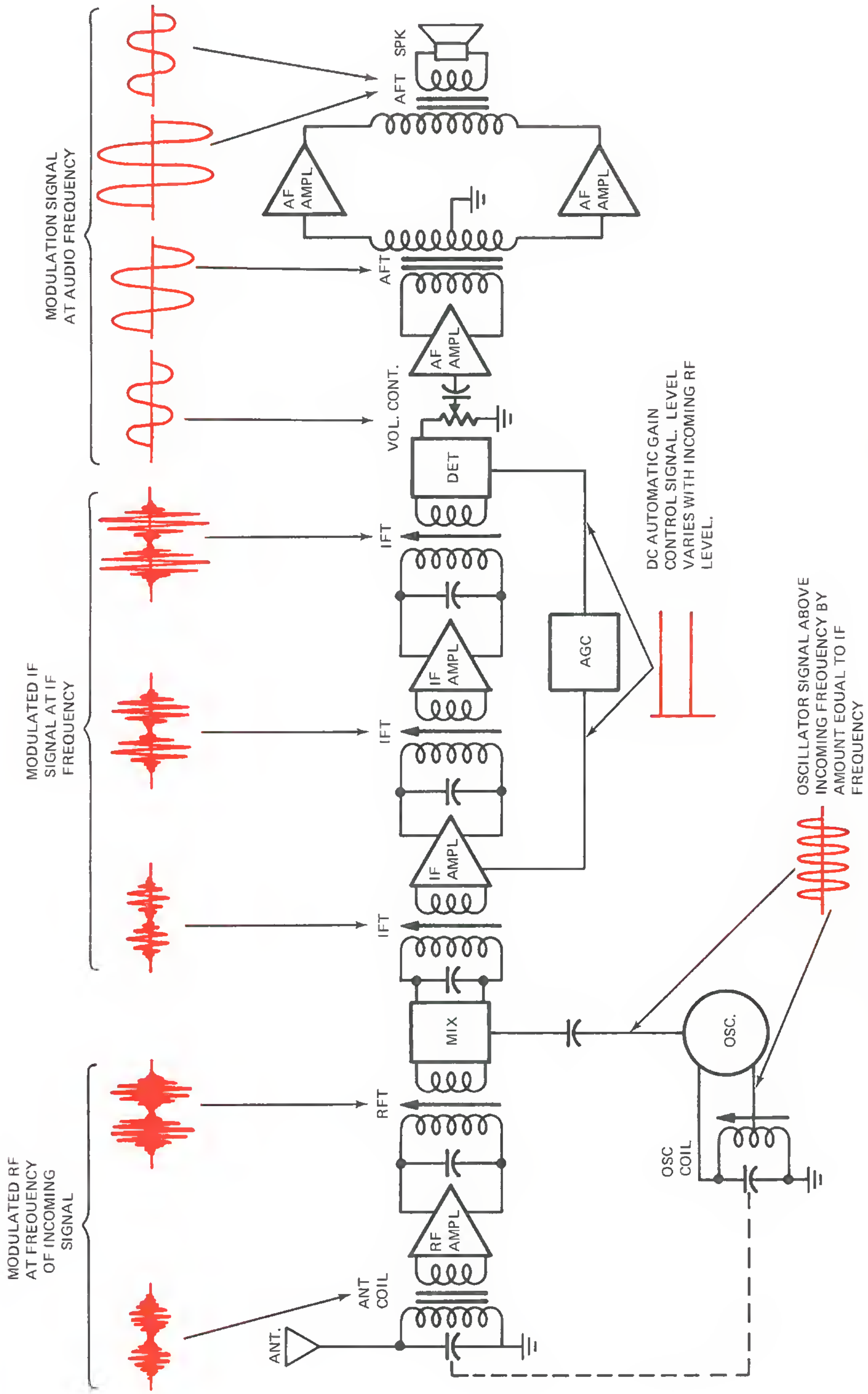


Fig. 8-1 Receiver Signals

signal, then we can conclude that at some point in the signal path there is a component which has failed in such a way as to completely block signal flow. Our problem then is to locate this single defective component. To better understand the troubleshooting process, let's use the circuit shown in figure 8-2 as an example.

The first step in any troubleshooting process is to check the DC operating potentials. In the example circuit, we would measure the value of V_{CC} and the V_{CE} of each transistor (in the case of an FET we measure V_{DS}, while for a tube we measure E_{PK}, E_{G2}, and observe filament operation). Normally, in a class A amplifier we expect V_{CE} to be approximately 1/2 V_{CC}. A class B amplifier, on the other hand, usually has a V_{CE} of about 9/10 V_{CC}.

As a result, we would expect V_{CE} to be about 6V for the first two transistors and about 10V for the two output transistors. If the V_{CE} value measured for one of the transistors was *greatly different* from the expected value, then we would suspect that stage of being defective.

After checking the operating potentials of the circuit, we apply a normal signal to the

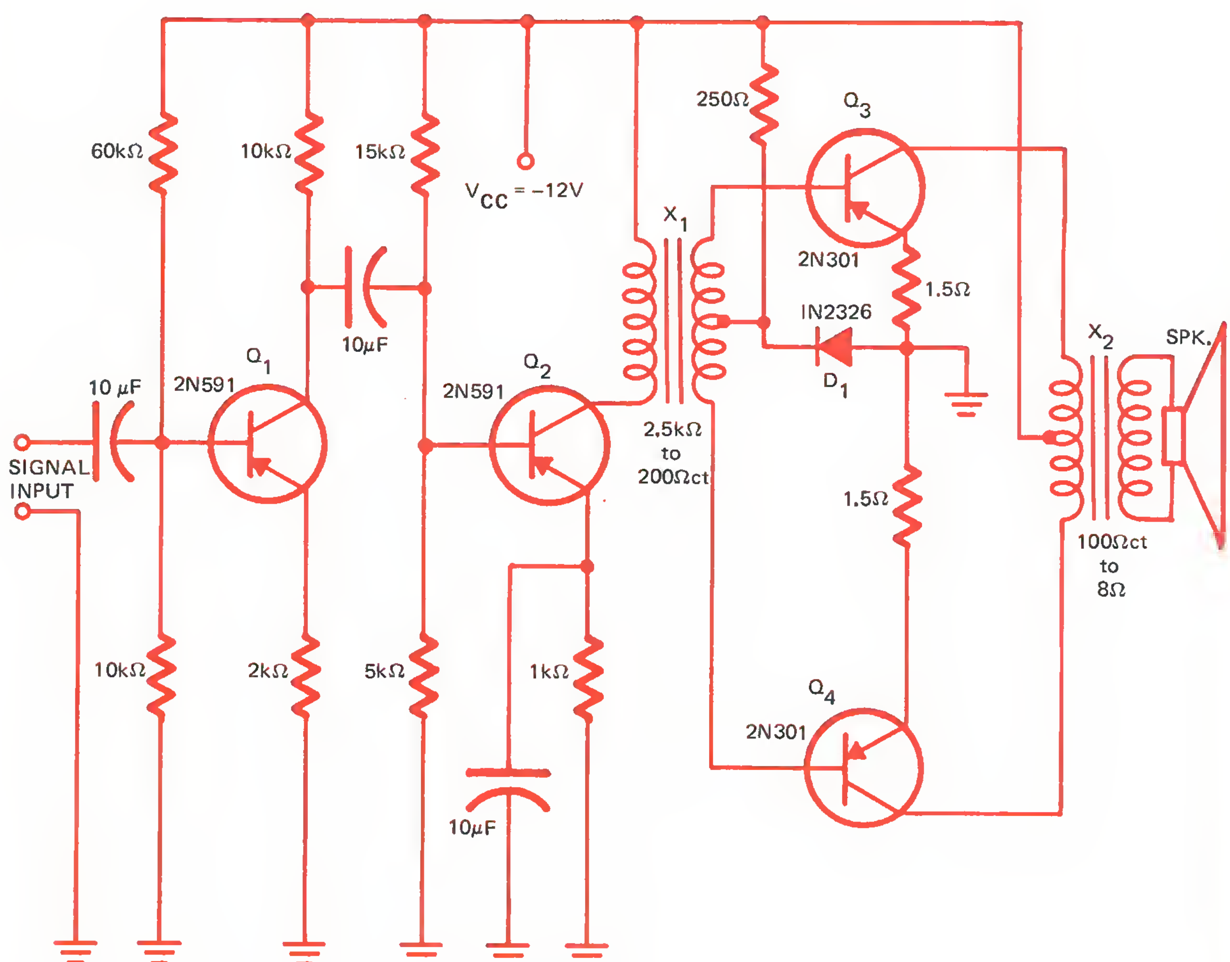


Fig. 8-2 A Typical Audio Section

input of the amplifier. We may then check the signal level (using an oscilloscope or other instrument) at various points within the circuit. By starting at the input and working toward the output, we can locate the point at which the signal becomes lost. In the example circuit, the signal should be checked at:

(a) the input terminals

(b) the base of Q_1

(c) the collector of Q_1

(d) the base of Q_2

(e) the collector of Q_2

(f) the base of Q_3

(g) the collector of Q_3

(h) the base of Q_4

(i) the collector of Q_4

(j) the output terminals

Notice that what we are doing here is checking the signal at the input and output of each device proceeding from the input to the output of the amplifier. When the point is found at which no signal is present, then we can usually isolate the defective component by a careful examination of the circuit. For instance, suppose that the signal is present at the base of Q_2 and not at the collector of Q_2. This being the case, we conclude that the defect must be caused by transistor Q_2, transformer X_1, or the resistor-capacitor network in the emitter circuit of Q_2. We can check each of these components one at a time to find the fault. Moreover, the operating potentials measured at Q_2 may help in locating the defect. If V_{CE} is equal to V_{CC}, then Q_2 must be an open circuit; or if V_{CE} is zero, then either Q_2 is shorted or something else (X_1 or the emitter resistor) is open. Measuring the DC voltage across the emitter resistor will indicate whether or not current is flowing in the circuit. If current is flowing, then there can be no open-circuited component.

In this way we can locate any component which is causing the signal to be lost.

REDUCED (or weak) OUTPUT SIGNAL. A seriously reduced output signal is usually caused by either a weak device (transistor with low h_{fe}, or low g_m in case of tubes and FET's) or by a defective coupling network. Checking the signal level at the input and output of each device will usually reveal the cause of the trouble. For instance if the signal is normal at the collector of Q_1 in the example circuit and severely reduced at the base of Q_2, then the coupling capacitor may have abnormally high reactance (low capacitance). This symptom could also be caused by a reduced input resistance (h_{ie}) for Q_2. However, a change in h_{ie} will normally cause a noticeable change in the DC operating potentials.

Transformers can fail causing a severe reduction (or even total loss) of signal. If one or more turns *of any winding becomes shorted, then the short circuit is reflected into all of the windings.* This trouble is difficult to identify and is usually the last possibility to be investigated. Perhaps the best way to test a transformer is to replace it with another known good one.

DISTORTED OUTPUT. Distortion in the output of an amplifier is most frequently caused by a shift in Q-point of one of the devices. Such a Q-point shift will normally be accompanied by a change in the operating potentials. The two most common causes of distortion are (a) defective devices (transistor, tubes, FET's), and (b) leaky coupling capacitors. The second of these (leaky capacitors)

causes Q-point shift by allowing DC current to leak from the output (collector) of one device to the input (base) of another.

It should be noted that output of a push-pull class B amplifier may become severely distorted if only one of the devices fails.

OSCILLATION. Perhaps the most troublesome amplifier problem is spurious oscillation. Oscillation occurs when a portion of the output of one stage gets into a previous input circuit and supports the input signal. Oscillation is therefore the result of regenerative feedback. Accidental regenerative feedback can occur when:

(a) circuit wiring is carelessly done

(b) circuit or component shielding is inadequate

(c) some component has signals from several stages applied to it

(d) the circuit gain has sharp peaks.

When undesirable oscillation occurs, the usual troubleshooting technique is to observe the oscillation with an oscilloscope while moving circuit components, wires, and shields. Any change in the oscillation waveform when a particular part is moved indicates that the part is one of the affected ones.

A fairly common cause of oscillation in an amplifier is inadequate power supply filtering. Such inadequate filtering allows signal current from all of the stages to flow through the power supply internal impedance thereby providing a feedback path.

Faulty ground connections may also cause oscillation by providing a resistance that is common to several stages.

NOISE AND HUM. Noise is randomly distributed energy usually produced by small disturbances within electronic components (resistors, transistors, capacitors, etc.). Carbon resistors and transistors are perhaps the worst offenders and the usual troubleshooting process is to watch the output noise level with an oscilloscope while suspected components are tapped briskly with a pencil or other light object. When a noisy component is located, it is replaced with a known good one. Noise is the one trouble which is frequently caused by more than one defective component at a time.

Hum is low frequency noise and is almost always related frequencywise to the 60-Hz AC line. Hum may get into an amplifier in one of four basic ways:

(a) The DC power source may be inadequately filtered allowing a hum signal to pass through into an amplifier stage.

(b) Ground loops (ground points which are at different AC levels) can induce hum into amplifier circuits.

(c) Any high impedance circuit can have hum induced into it by stray magnetic fields from transformers, power lines, etc.

(d) Vacuum tubes with AC filament voltages are particularly bad about inducing hum into amplifier circuits.

In troubleshooting excessive hum in an amplifier, the four possibilities named above are examined one at a time, usually in the order given.

If the audio section is working normally but the receiver is still dead, then we move back to the input of the first IF stage and apply a modulated test signal at the IF frequency (usually about 455 kHz). If we hear the test tone in the speaker then the IF am-

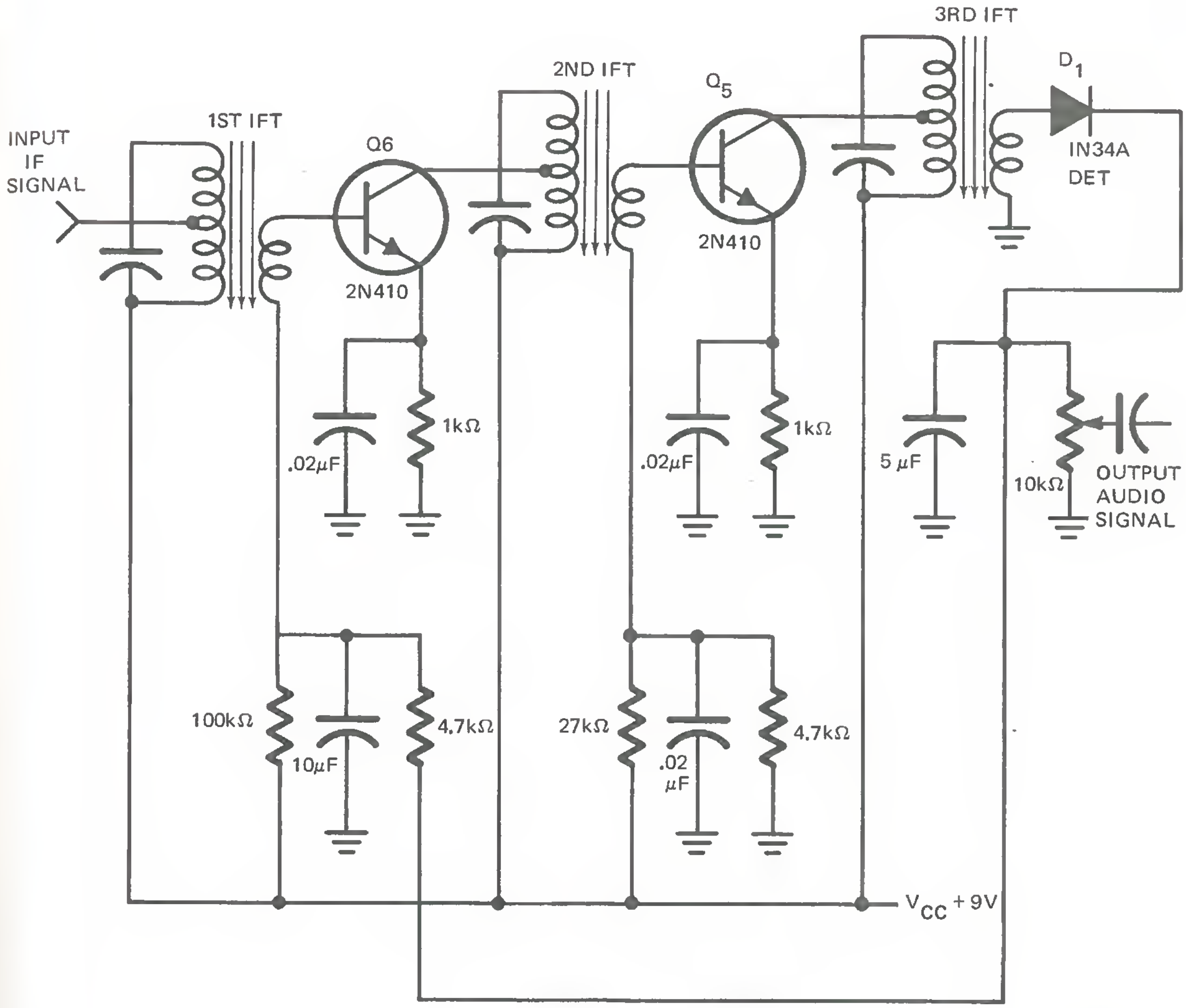


Fig. 8-3 A Typical IF Section with Detector

plifiers, detector and audio amplifiers are working. If there is not a normal test tone output, then the trouble must be in the IF amplifiers or in the detector.

We would therefore proceed to troubleshoot these stages. Figure 8-3 shows a typical IF and detector circuit. The troubleshooting procedure is substantially the same as for a transformer-coupled audio amplifier. That is, we usually start out by measuring V_{CE} of each transistor. It should be about 90% of V_{CC} (or 8 volts in figure 8-3). Any value *greatly different* from this would indicate an abnormally operating stage.

In tracing signal flow through an IF section it is a good idea to connect an audio indicator (oscilloscope or VOM) across the detector output. Then inject a modulated IF signal at the collector of the last IF amplifier (Q_5). You will see the audio on the indicator and hear it in the speaker if the 3rd IFT and detector stage are ok.

In some cases the signal generator lead wire will detune the IFT enough to prevent signal passage. This can be prevented by using a 10 kΩ resistor in series with the generator lead.

If the 3rd IFT and detector are working move the signal injection back to the base of Q_5 then to the collector of Q_6. If the signal goes through in both cases, move back to the base of Q_6 then to the input of the 1st IFT. When a signal stoppage is encountered, check out the circuit between the point where the signal is injected and the previous injection point.

The most common IF section problems are:

1. Loss of signal due to a defective transistor, diode, IFT, or to improper IF alignment.
2. Oscillation due to improper alignment or a defective transistor. (Oscillation can also occur when the chassis is improperly shielded such as when it is removed from its case.)
3. Distortion caused by a bad transistor or a shorted turn in an IFT.

If the IF section is working properly but the set is still dead, the trouble must be in the RF mixer, or oscillator circuits.

The usual procedure is to apply a modulated *IF* signal to the antenna. If the IF signal goes through from antenna to speaker, then the trouble must be in the oscillator or mixer stage.

If the IF signal does not go through from antenna to speaker, then the trouble must be in the RF amplifier or mixer stage.

If the IF signal does go through, the next step is to put in a modulated RF signal. Then tune the receiver to the signal frequency and check the oscillator output. This can perhaps best be done with an oscilloscope.

When a defective RF mixer, or oscillator stage is identified, then troubleshooting is accomplished by measuring circuit voltages to pinpoint the defective component.

MATERIALS

- 1 AM radio receiver
- 1 Circuit diagram for the receiver
- 1 RF signal generator
- 1 Oscilloscope
- 1 VOM or FEM
- 1 Nonmetallic hex type tuning tool
- 1 Nonmetallic screw type tuning tool

PROCEDURE

1. Turn on all of your equipment and allow several minutes for it to warm up.
2. Check the general operation of the receiver to insure that it is working.
3. Check the IF, oscillator, and RF alignment to insure that they are correct.
4. When you have completed step 3 tell your instructor. He will put a trouble in your receiver.

5. When your trouble has been installed, carry out the following:
   (a) Sectionalize the trouble to the audio, IF, or RF section. Record the tests you make, the results of the test, and your conclusion. For example:

       Test — Audio signal across volume control
       Results — No sound from speaker
       Conclusion — Trouble in audio section

   (b) Localize the trouble to a particular stage by measuring voltages or by test signal injection. Again record your test, results, and conclusion. For example:

       Test — Measure V_{CE} of first audio stage
       Results — V_{CE} is zero
       Conclusion — Defect in first audio stage

   (c) Isolate the defect within the stage by making voltage and resistance measurements. Record each test you make, the results of the test, and your conclusion. For example:

       Test — Measure voltage across emitter resistor in first audio stage
       Results — Voltage is nearly equal to V_{CE}
       Conclusion — Transistor is shorted or saturated
       Test — Check transistor
       Results — Shorted emitter to collector
       Conclusion — Should replace transistor

6. When you have located the defect, tell your instructor. He will verify your conclusion and then put another trouble in your receiver.

7. Repeat steps 5 and 6 as many times as possible during the lab period.

ANALYSIS GUIDE. Using your own words write a procedure for sectionalizing, localizing, and isolating troubles in a superheterodyne receiver. List the equipment you would need to service such a receiver and tell what each piece of equipment is used for.

PROBLEMS

1. The audio section shown in figure 8-2 is dead. V_{CE} at Q_1 is zero and the voltage across the emitter resistor of Q_1 is zero. What component failures could cause this?

2. V_{CE} of all transistors in figure 8-2 is zero. Where would you look for the trouble?

3. There is no output from figure 8-3 when an IF signal is applied to the collector of Q_5. What would you do next?

4. V_{CE} of Q_6 in figure 8-3 is +9 volts while V_{CE} of Q_5 is 7.6 volts. What components could have failed? Explain how each failure could cause these voltages.

5. The receiver shown in figure 8-4 is dead. An IF signal applied to the antenna can be clearly heard at the speaker. What could be the trouble?

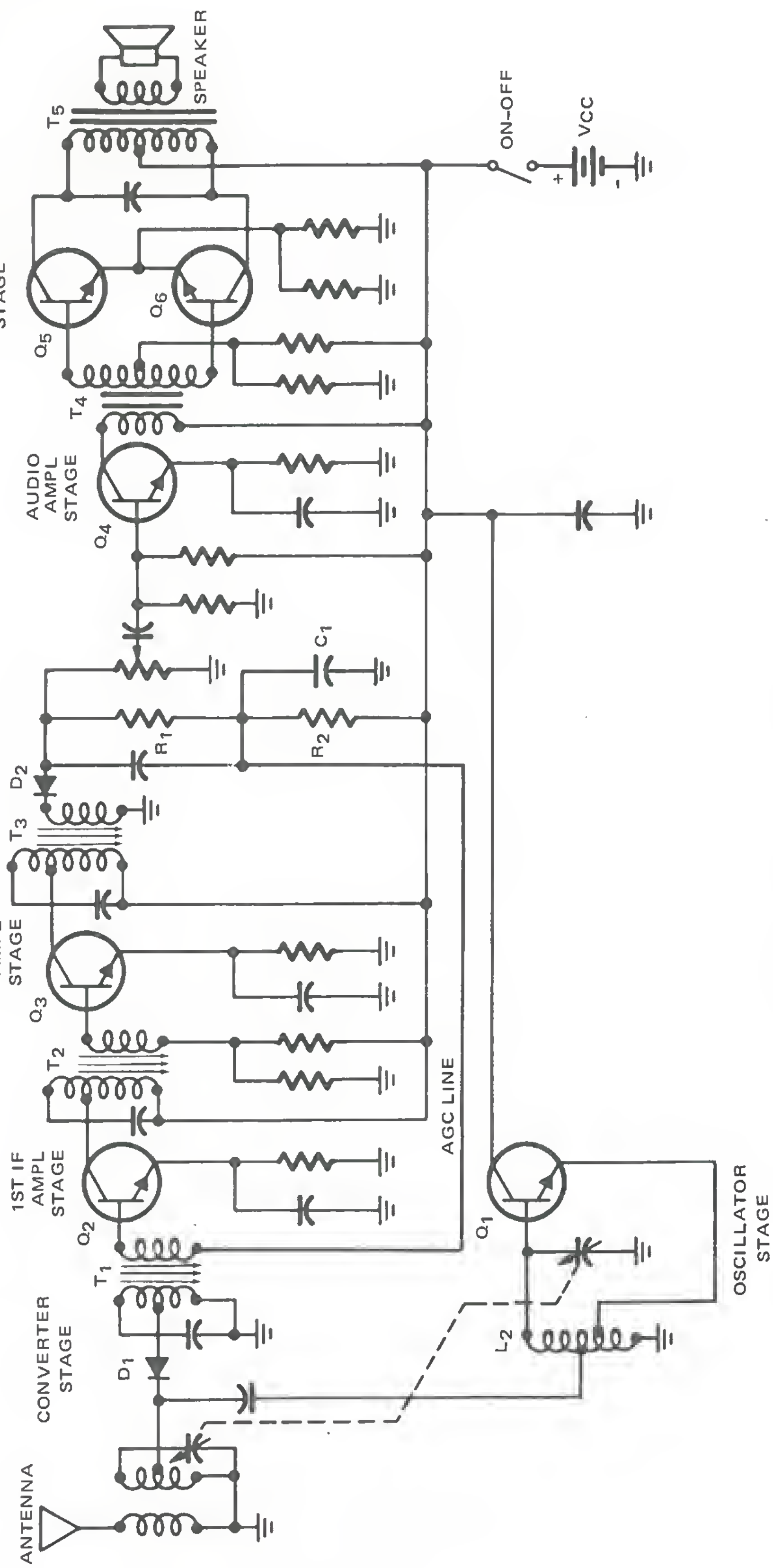


Fig. 8-4 A Superheterodyne Receiver Circuit

experiment 9 RF AMPLIFIERS

INTRODUCTION. RF amplifiers are perhaps the fundamental circuits of all radio communication systems. In this experiment we will examine the operating characteristics of some typical types of RF amplifier circuits.

DISCUSSION. Radio frequency amplifiers may be either linear amplifiers (class A) or nonlinear amplifiers (class B or C). Examples of class A (linear) RF amplifiers are the RF and IF stages in the superheterodyne receiver shown in figure 9-1. These amplifiers are biased by the emitter and base resistors so that the collector current flows during the entire 360 degrees of the input signal cycle.

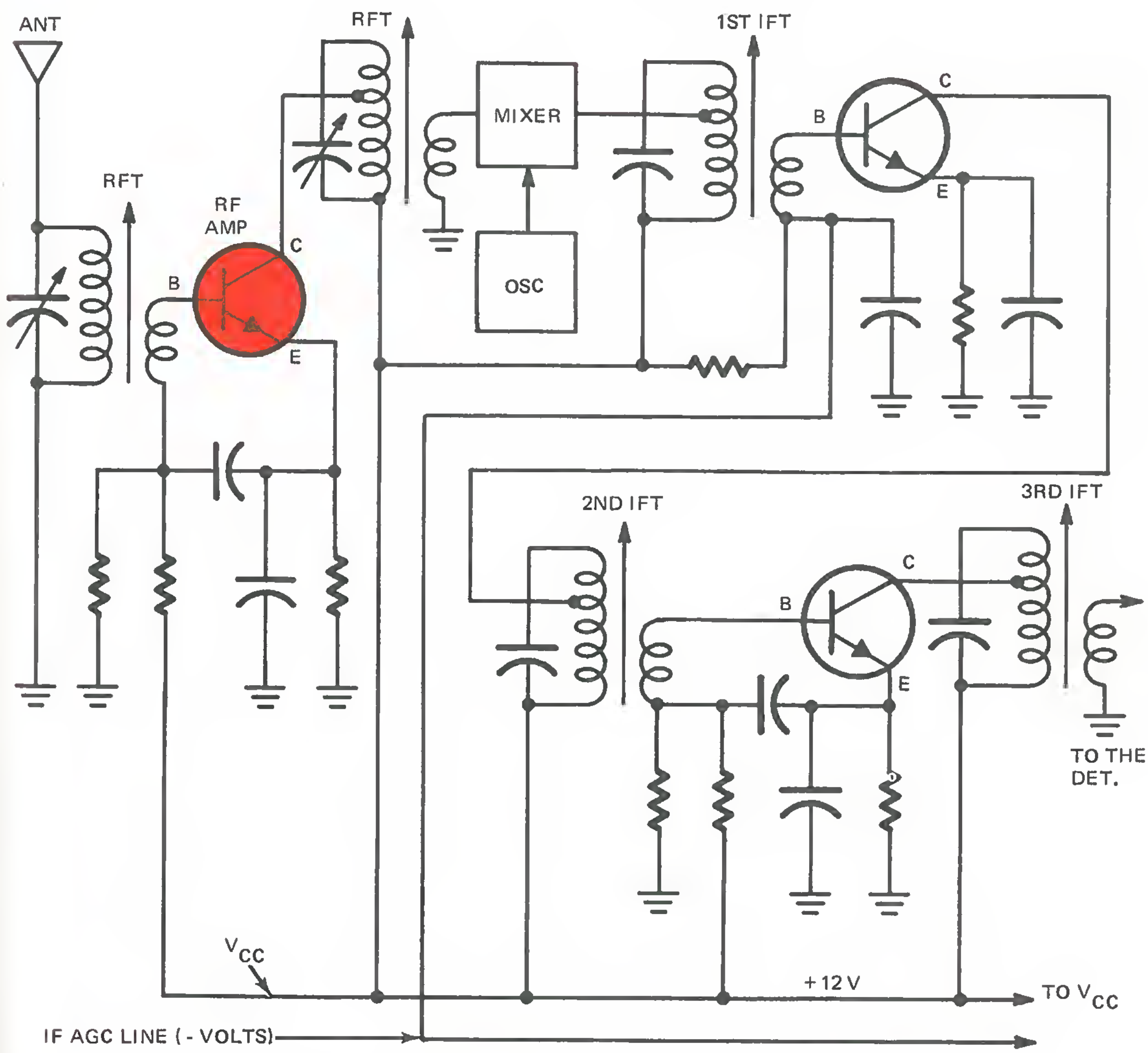


Fig. 9-1 Class A RF Amplifiers in a Receiver

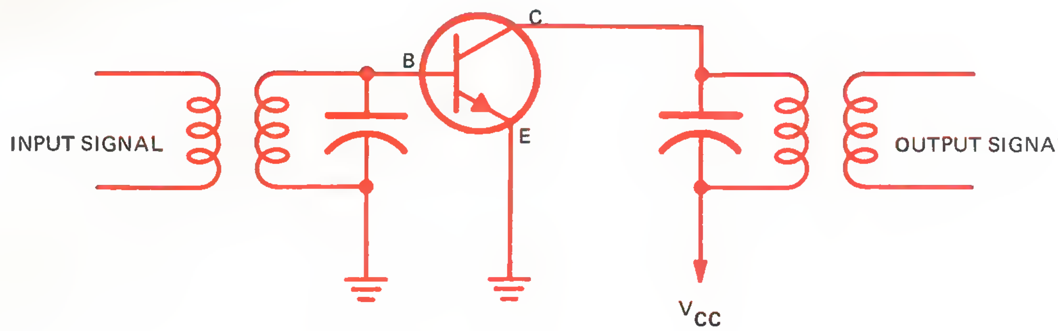


Fig. 9-2 A Nonlinear RF Amplifier

The only major difference between a linear RF amplifier and an audio amplifier lies in the collector loading circuit. The RF amplifier uses a tuned tank circuit where an audio amplifier uses an untuned load.

Nonlinear RF amplifiers, on the other hand, are a somewhat different story. Let's consider the amplifier shown in figure 9-2. The input transformer is single tuned to the incoming signal frequency. Let's suppose that the voltage across the secondary of the input transformer has the form

$$e_{BE} = E_s \cos \omega_s t \tag{9.1}$$

The input characteristic of the transistor is something like figure 9-3. We can represent such a characteristic by the power series

$$i_B = A_0 + A_1 e_{BE} + A_2 e_{BE}^2 + A_3 e_{BE}^3 + \ldots + A_n e_{BE}^n$$

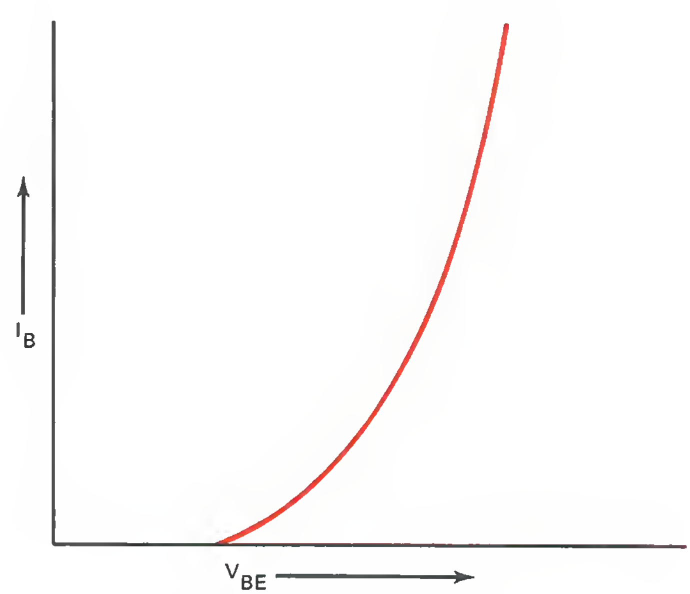


Fig. 9-3 A Typical Transistor Input Curve

In most practical cases the first two terms A_0 and $A_1 e_{BE}$ will be so small that they can be dismissed. Also, it is rare for the input to be large enough for the fourth degree term ($A_4 e_{BE}{}^4$) and beyond to be important. Consequently, we can usually approximate the base current effectively by

$$i_B = A_2 e_{BE}^2 + A_3 e_{BE}^3 \qquad (9.2)$$

If we substitute our input voltage into this expression, we will have

$$i_B = A_2 E_s{}^2 \cos^2 \omega_s t + A_3 E_s{}^3 \cos^3 \omega_s t$$

Then using the trigonometric identities

$$\cos^2 \omega_s t = \frac{1}{2}(1 + \cos 2\omega_s t)$$

and

$$\cos^3 \omega_s t = \frac{1}{4}(3 \cos \omega_s t + \cos 3\omega_s t)$$

we have

$$i_B = \frac{A_2 E_s{}^2}{2} + \frac{3A_3 E_s{}^3}{4} \cos \omega_s t + \frac{A_2 E_s{}^2}{2} \cos 2\omega_s t + \frac{A_3 E_s{}^3}{4} \cos 3\omega_s t \qquad (9.3)$$

Now since the collector current (i_C) is approximately

$$i_C = h_{fe} i_B$$

We can write the collector current expression as

$$i_C = \frac{h_{fe} A_2 E_s{}^2}{2} + \frac{3h_{fe} A_3 E_s{}^3}{4} \cos \omega_s t + \frac{h_{fe} A_2 E_s{}^2}{2} \cos 2\omega_s t + \frac{h_{fe} A_3 E_s{}^3}{4} \cos 3\omega_s t$$

To simplify things let's notice that each of the four coefficients has a different value. Let's call them I_C, I_1, I_2, and I_3 respectively. If we do this we have

$$\boxed{i_C = I_C + I_1 \cos \omega_s t + I_2 \cos 2\omega_s t + I_3 \cos 3\omega_s t} \qquad (9.4)$$

as our expression for the collector current.

Let's examine these four terms one at a time. The first term (I_C) is a DC collector current component. The second term ($I_1 \cos \omega_s t$) is a fundamental component having the same frequency as the input signal. The third term ($I_2 \cos 2\omega_s t$) is a second harmonic term having twice the input frequency. The fourth term ($I_3 \cos 3\omega_s t$) is a third harmonic signal having three times the frequency of the input signal.

These four collector current terms present a number of interesting possibilities. The DC term could be used as an AGC signal. While this is rarely done, it is an interesting possibility.

The fundamental frequency term ($I_1 \cos \omega_s t$) suggests that we can use the circuit in figure 9-2 simply to amplify radio frequency signals. To do this we simply tune the collector tank to the same frequency as the input signal. This type of operation is very much like that of a linear RF amplifier.

The second harmonic term of the collector current suggests something that we have not considered before. It would appear that we can put in a signal of one frequency and take out a signal which has twice that frequency. This is, in fact, the case. All we need do is tune the collector tank to twice the input frequency. When this is done the output voltage is usually somewhat less than

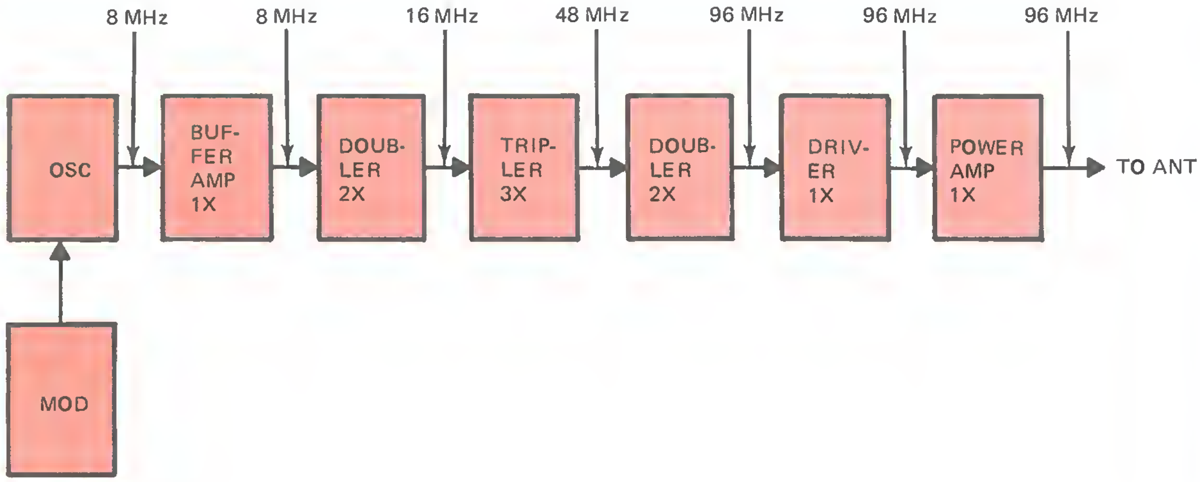


Fig. 9-4 An FM Transmitter

when the amplifier output is tuned to the fundamental frequency. An RF amplifier used in this way is called a *frequency doubler.*

The presence of the third harmonic term suggests that we might also use the amplifier as a *frequency tripler.* And sure enough, if we tune the output tank to three times the input frequency, we get frequency tripling action. The output level again is reduced as compared to a doubler or fundamental frequency amplifier.

Amplifiers which are used as doublers or triplers are classified as *frequency multipliers* and are used quite a lot in high frequency transmitters. The system in figure 9-4 shows how an 8 MHz oscillator can be used to produce a 96 MHz output for an FM transmitter.

The amplifier circuit we have considered so far is of class B operation. This class of operation is sometimes used when amplitude modulation is present on the RF signal being processed.

With unmodulated RF signals, class C amplifiers are more commonly used because they can be made to operate more efficiently. However, since class C operation requires that the amplifier be biased beyond cutoff, they tend to distort amplitude modulation. Figure 9-5 shows a comparison of class A, B, and C operation. Notice that with class A operation the base current and, consequently, collector current flows during the entire input cycle (subject to saturation and cutoff limits). Moreover, the base current is relatively linear. In the class B condition the transistor operates with virtually zero bias. As a result, base current flows only during one half of the input cycle. The resulting collector current is then in the form of half cycle pulses which contain substantial amounts of the second and third harmonics.

With class C operation the base is biased well beyond cutoff. So, base current can only flow during that small part of a cycle when the signal carries the base out of cutoff. The consequent collector current pulses are very rich in second and third harmonic content.

The problem associated with using class C amplifiers with amplitude modulated input

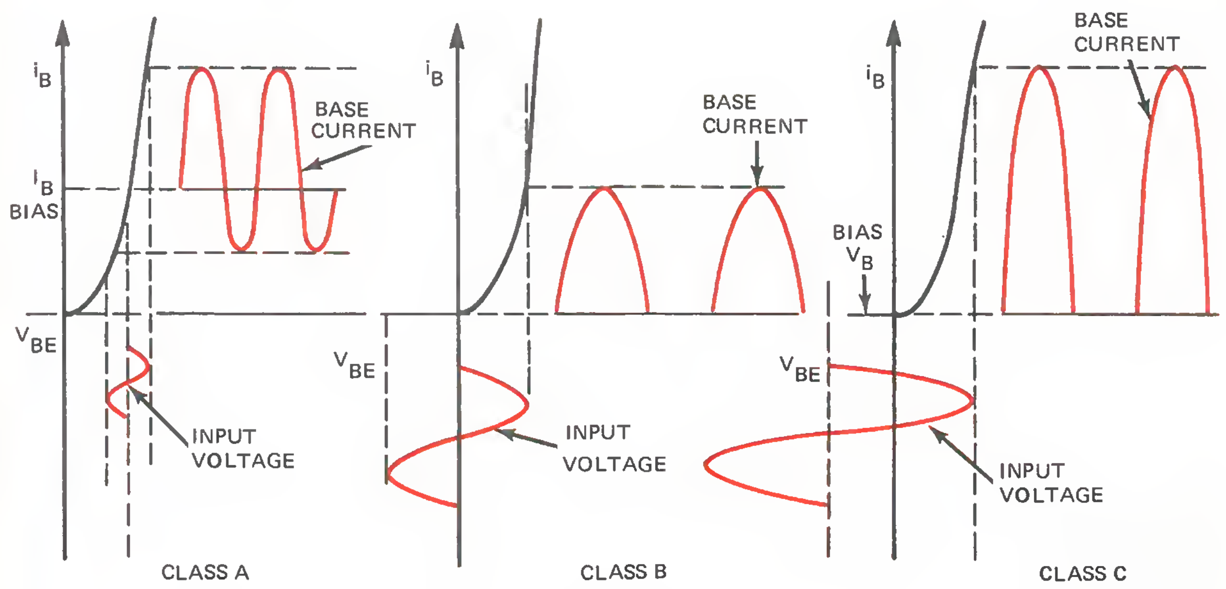


Fig. 9-5 Class A, B, and C Input Characteristics

signals can be visualized by comparing the three classes of operation. Figure 9-6 shows input characteristics of each class with an AM signal. The class A amplifier handles the entire modulated signal relatively linearly. The class B amplifier has a base current which preserves one half of the modulation envelope fairly well. The individual RF half-cycles excite the collector tank causing it to *ring*. This ringing within the tank tends to

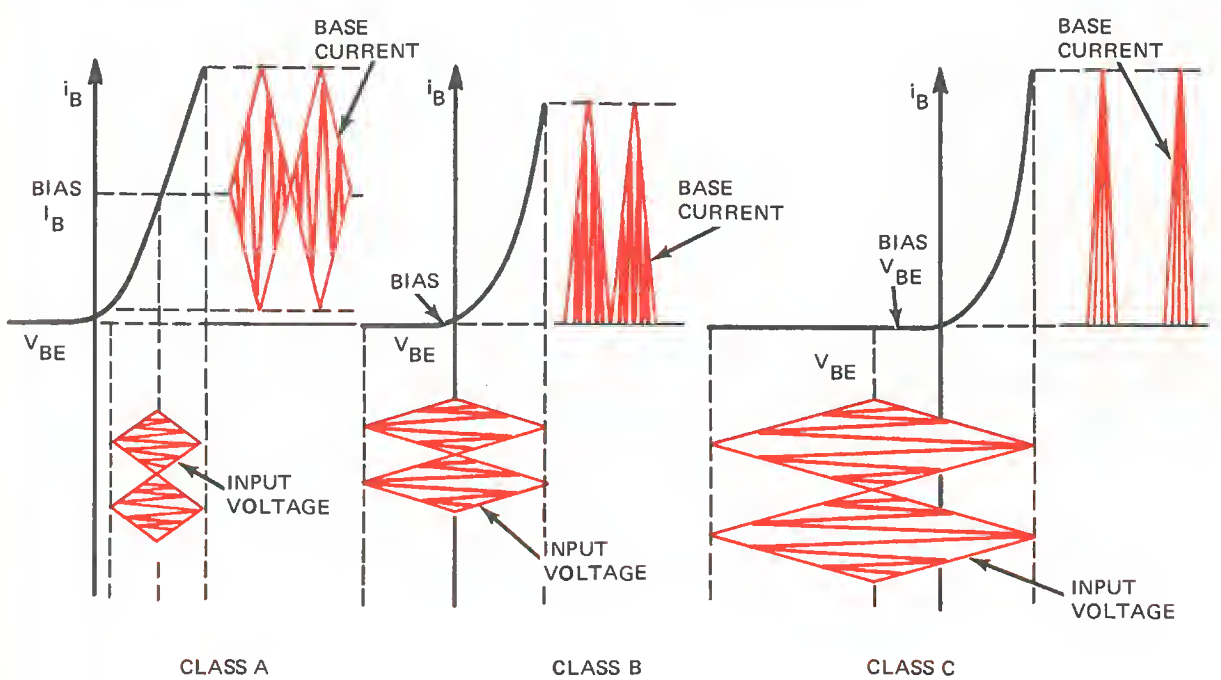


Fig. 9-6 Class A, B, and C with AM Inputs

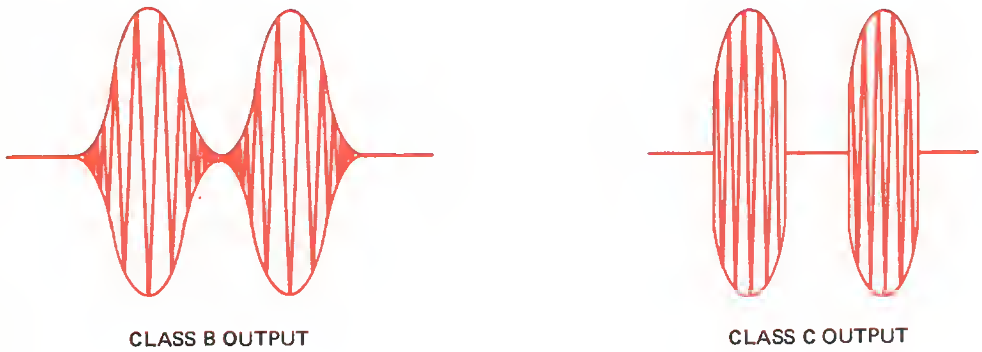


Fig. 9-7 Outputs of Class B and C Amplifiers

fill in the missing half-cycles so that the class B *output* looks about like figure 9-7. While there is *some* modulation distortion with a class B amplifier it is not nearly so severe as with a class C amplifier.

The overall result is that class C amplifiers are usually not used for handling amplitude modulated signals. Perhaps the principal exception is when the modulation is in the form of rectangular pulses such as in CW telegraphy. With rectangular pulse modulation, class C operation does not appreciably interfere with the modulation waveform.

It should not be assumed that class C amplifiers can not be used at all in linear AM service. They are, in fact, often used. But the modulation is performed *after* amplification. Figure 9-8 shows a typical AM transmitter block diagram. All of the amplifiers in this transmitter could be class C circuits.

Normally in class C operation the bias which holds the amplifier cut off during the no output period is provided by a separate bias power supply. Figure 9-9 shows a simple class C bias arrangement.

Radio frequency amplifiers of all classes may be made using vacuum tubes as well as transistors. However, with the rapid im-

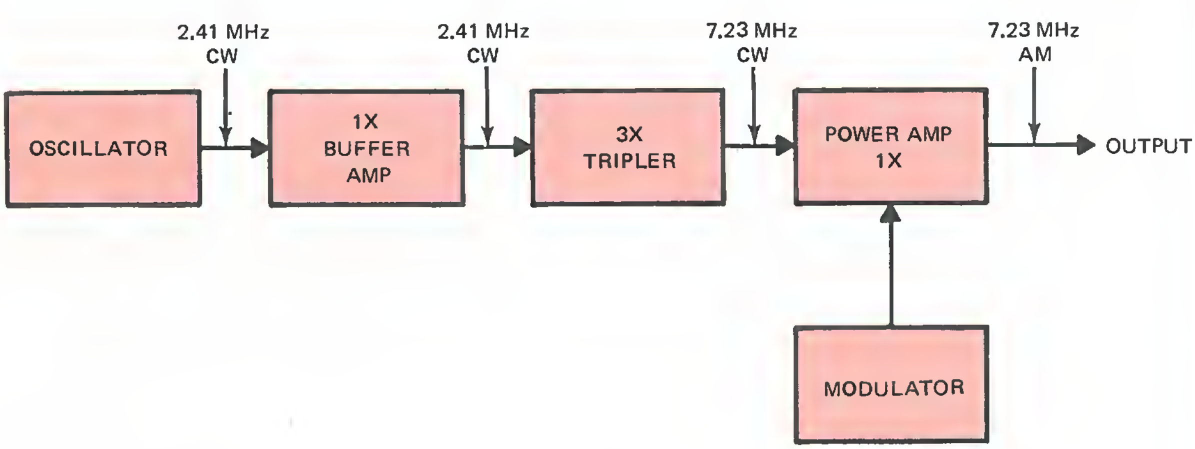


Fig. 9-8 An AM Transmitter

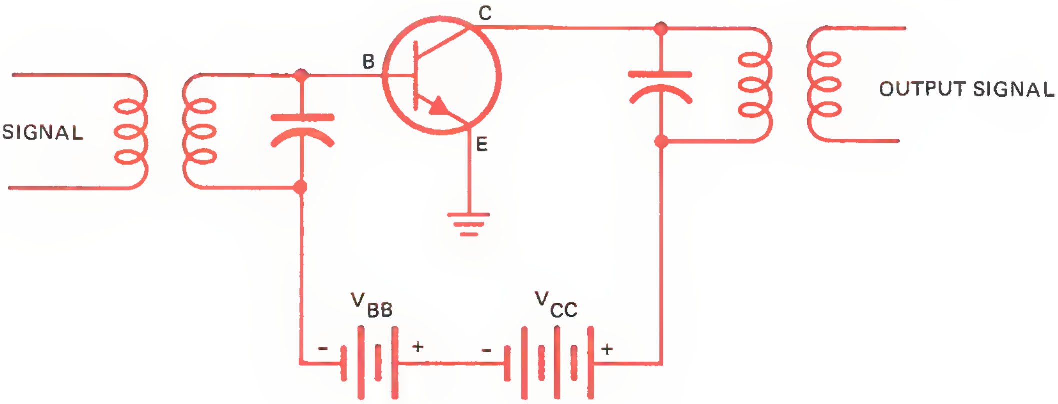


Fig. 9-9 A Class C Amplifier Circuit

provements being achieved in high frequency transistors, vacuum tube amplifiers are being encountered less and less.

In this experiment we will use an AM transmitter with the circuitry as shown in figure 9-10. In this transmitter circuit V_1 is

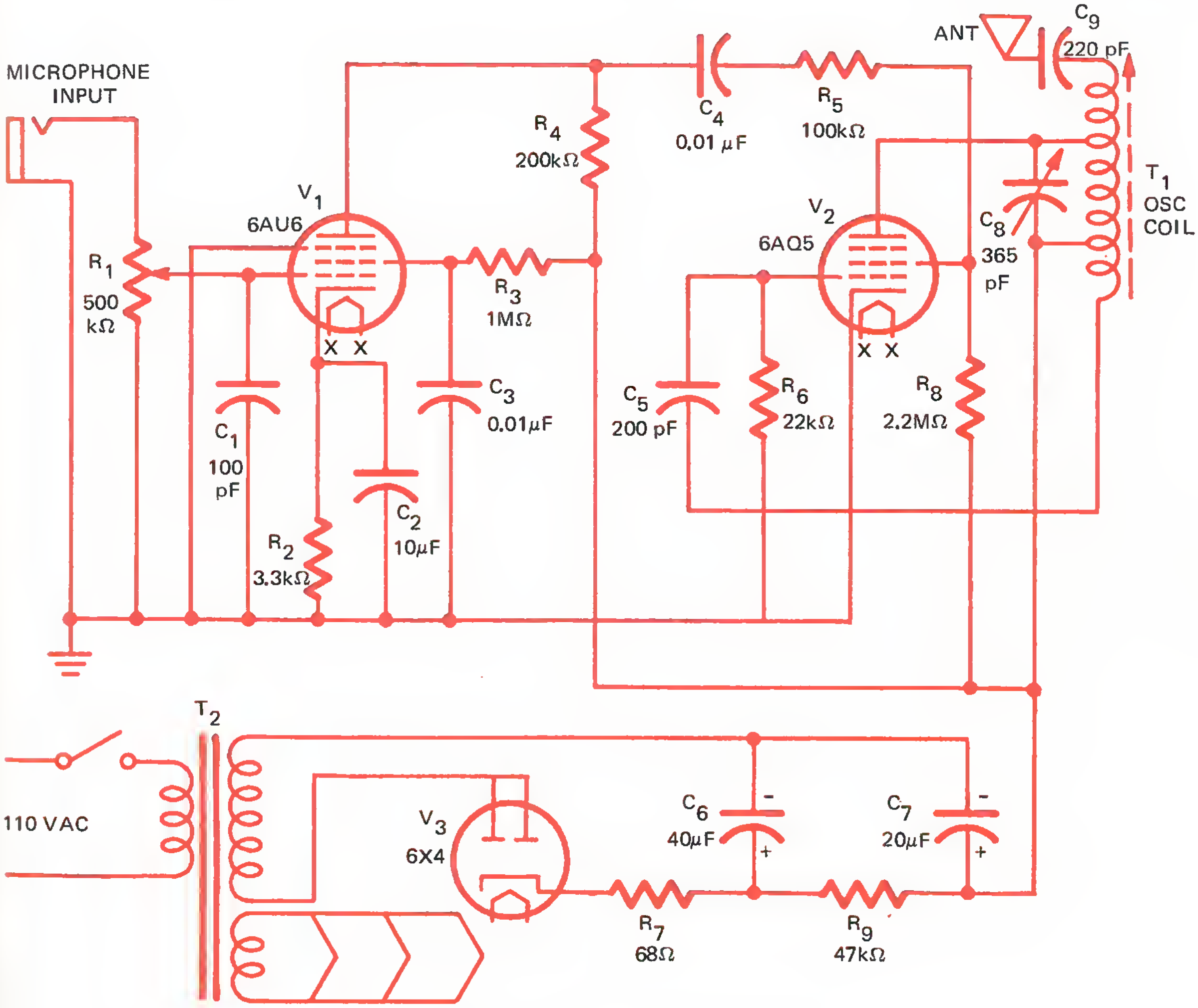


Fig. 9-10 The AM Transmitter

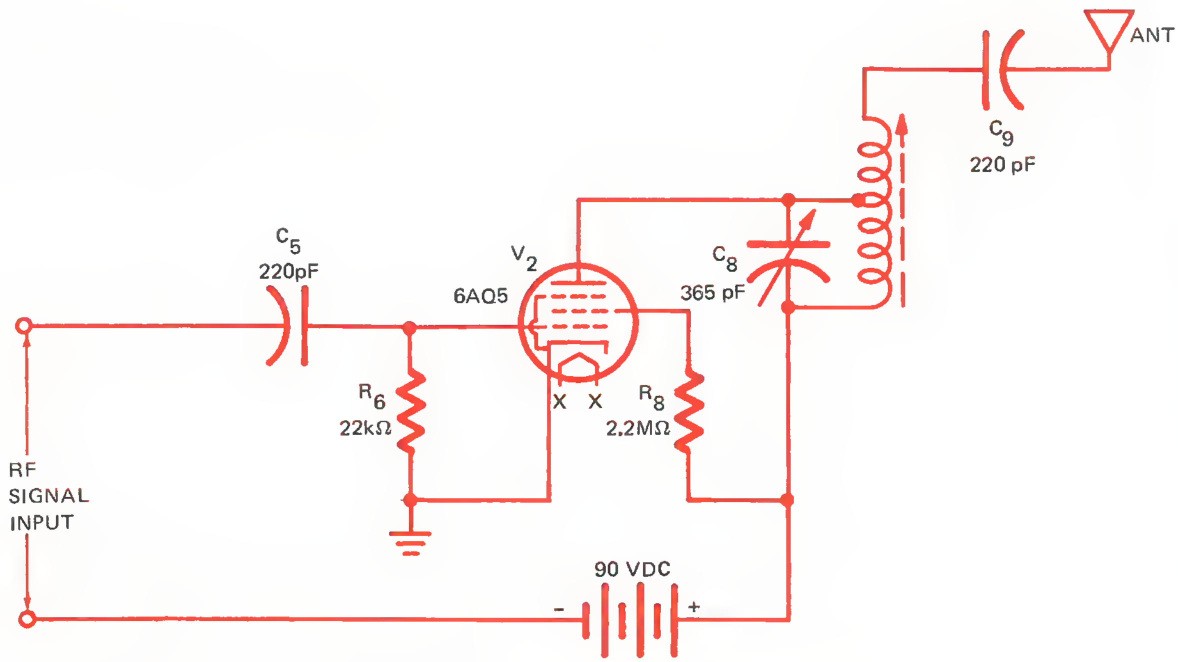


Fig. 9-11 The Experimental RF Amplifier

an audio amplifier, V_2 is an RF oscillator and V_3 is the power supply rectifier. We are going to convert the RF oscillator (V_2) into an RF amplifier for the purposes of this experiment. To do this you must do two things: first, unplug the microphone and turn the volume control down to minimum; second, locate C_5 (220 pF) and disconnect the end which goes to the oscillator coil. In effect we now have an RF amplifier with a circuit diagram like figure 9-11. We are now ready to carry out the experiment.

MATERIALS

- 1 AM radio transmitter
- 1 RF signal generator
- 1 Oscilloscope
- 1 Nonmetallic screw type tuning tool

PROCEDURE

1. Turn on all of your equipment and allow it to warm up.
2. Screw the tank coil slug in until it is almost fully into the coil.
3. Set the tank capacitor to its minimum capacitance setting.
4. Connect the oscilloscope across the tank circuit. Use a low capacity or properly compensated x10 probe.
5. Set the RF signal generator for its maximum output at about 1 MHz. Set the generator modulation control for an *unmodulated* signal.
6. Connect the generator across the input of the RF amplifier.
7. Tune the generator for the maximum amplifier output in the vicinity of 1 MHz. Sketch the output waveform. Then record its peak-to-peak amplitude (E_1) and frequency (f_1).

DATA WITH MODULATED INPUT

INPUT DATA		OUTPUT DATA
E_s = ______ f_s = ______	A = ______	E_1 = ______ f_1 = ______
E'_s = ______ f'_s = ______		E_2 = ______ f_2 = ______
E''_s = ______ f''_s = ______		E_3 = ______ f_3 = ______

Fig. 9-12 The Data Tables

DATA WITH UNMODULATED INPUT

INPUT DATA	OUTPUT DATA
E_s = ________ f_s = ________	E_1 = ________ f_1 = ________
E'_s = ________ f'_s = ________	E_2 = ________ f_2 = ________
E''_s = ________ f''_s = ________	E_3 = ________ f_3 = ________

A_V = ________

Fig. 9-12 The Data Tables (cont'd)

8. Move the oscilloscope so that you can view the generator output while it is connected to the amplifier. Sketch the input waveform. Then record the peak-to-peak amplitude (E_s and frequency f_s).
9. Calculate the stage voltage gain (A_V).
10. Return the oscilloscope to the previous position across the tank circuit.
11. Tune the signal generator down to the frequency that is 1/2 f_s. Then tune finely for the maximum amplifier output. Sketch the output waveform. Then record the E_2 and f_2 as in step 7.
12. Repeat step 8. Record the values as E_s' and f_s'. Then repeat step 10.
13. Tune the signal generator down to the frequency that is 1/3 f_s. Then tune finely for the maximum amplifier output. Sketch the output waveform. Then record E_3 and f_3.
14. Repeat step 8. Record the values as E_s'' and f_s''. Then repeat step 10.
15. Switch on the signal generator modulation. Then repeat steps 7 through 14, but this time view the modulation envelope and record its waveform, peak-to-peak voltage, and frequency.
16. Restore the experimental circuit to the condition it was in originally.

ANALYSIS GUIDE. In analyzing your results from this experiment you should consider the class of operation of the amplifier in relationship to its operation. Specifically, did the circuit perform as a frequency multiplier in a manner appropriate for its class? Did it handle a modulated signal in the manner you expected? Explain any observed differences between the input and output waveforms.

PROBLEMS

1. Explain in your own words the differences between the circuits of a class A, B, and C RF amplifier.
2. What *circuitry* differences would you expect between a doubler and a tripler?
3. Discuss the input characteristics of a vacuum tube amplifier using an approach similar to figure 9-6.
4. The transmitter in figure 9-13 has a crystal oscillator operating at 1.416 MHz. What is the output frequency?

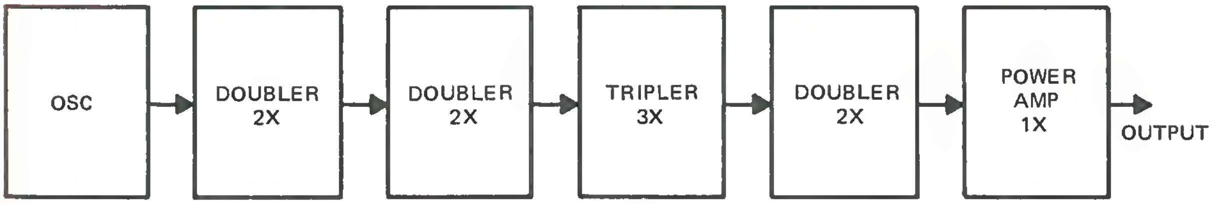


Fig. 9-13 Transmitter for Problems 4, 5, 6, and 7.

5. If the oscillator frequency drifts downward 2 kHz, what will be the new output frequency?

6. Each of the stages in figure 9-13 has an output tank circuit. To what frequency is each of these tanks tuned?

7. The transmitter in figure 9-13 is to be frequency modulated. The output is to vary ±25 kHz from its center frequency. What is the corresponding oscillator frequency variation?

8. Can a square law device that has the characteristic

$$i = Ae^2$$

be used as a frequency doubler? Explain your answer.

9. Can the device in problem 8 be used as a frequency tripler? Explain your answer.

experiment 10 *AMPLITUDE MODULATION*

INTRODUCTION. Information is conveyed from one point to another in a radio communications system by a process called modulation. In this experiment we will examine the process of amplitude modulation and some of the circuits used to produce it.

DISCUSSION. Amplitude modulation is a process whereby a relatively low-frequency modulation signal is superimposed upon a relatively high-frequency RF carrier.

You should understand, however, that *modulation does not occur* when the two signals are mixed linearly. To illustrate this point let's consider the linear mixing circuit shown in figure 10-1. The characteristic equation of this circuit is

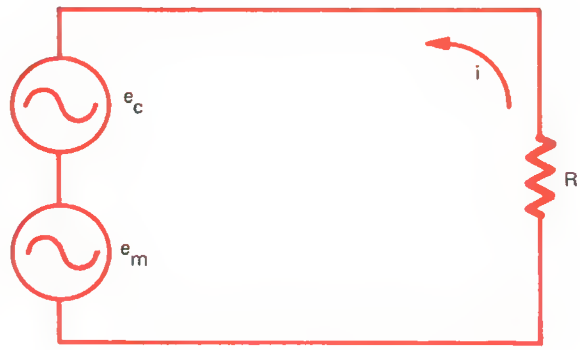


Fig. 10-1 A Linear Mixing Circuit

$$i = \frac{1}{R} e_T$$

And since $e_T = e_c + e_m$ we have

$$i = \frac{1}{R} e_c + \frac{1}{R} e_m$$

Then if the two input voltages have the forms

$$e_c = E_c \cos \omega_c t$$

and

$$e_m = E_m \cos \omega_m t$$

the circuit current will be

$$\boxed{i = \frac{1}{R} E_c \cos \omega_c t + \frac{1}{R} E_m \cos \omega_m t} \qquad (10.1)$$

Notice that the total current is simply the sum of the currents that would result from each signal independently. *This is not modulation, it is mixing.*

If we were to apply these two mixed signals to an antenna, the carrier signal would radiate from the antenna but the modulation signal would not. A remotely located receiver would receive only the carrier signal but not the modulation. It is worth noting that if we observed the current waveform on an oscilloscope it would look like figure 10-2 if the

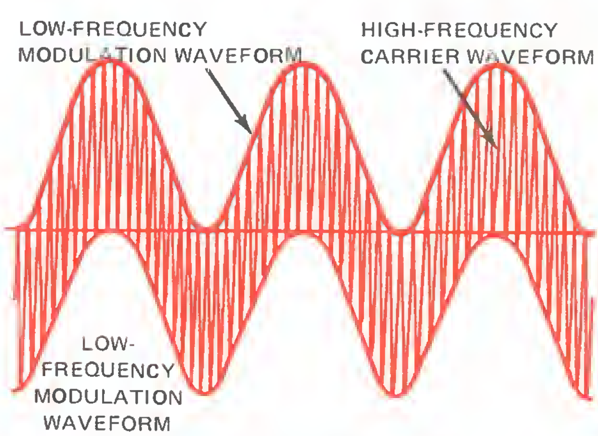


Fig. 10-2 Linearly Mixed Signals at a Slow Sweep Rate

scope is sweeping slowly enough to view the modulation waveform. This waveform is characteristic of linear mixing when there is considerable difference in the frequencies of the signals. I'm sure you have observed such a waveform from time to time.

Actual modulation occurs only when the two signals are mixed *nonlinearly*. Let's consider a very simple case first. Figure 10-3 shows our two signals applied to a diode. The characteristic equation of the diode we can take as approximately

$$i = A_1 e_T + A_2 e_T^{\,2}$$

And since e_T is still equal to $e_c + e_m$ we have

$$i = A_1(e_c + e_m) + A_2(e_c + e_m)^2$$

or in this case

$$i = A_1 E_c \cos \omega_c t + A_1 E_m \cos \omega_m t + A_2(E_c \cos \omega_c t + E_m \cos \omega_m t)^2$$

Squaring the trigonometric binomial as indicated gives us

$$\begin{aligned} i = {} & A_1 E_c \cos \omega_c t + A_1 E_m \cos \omega_m t \\ & + A_2 E_c^{\,2} \cos^2 \omega_c t \\ & + 2A_2 E_c E_m \cos \omega_c t \cos \omega_m t \\ & + A_2 E_m^{\,2} \cos^2 \omega_m t \end{aligned}$$

Then if we simplify using the trigonometric identities

$$\cos^2 \omega t = \frac{1}{2} + \frac{1}{2} \cos 2\omega t$$

and

$$\cos \omega_c t \cos \omega_m t = \frac{1}{2} \cos(\omega_c + \omega_m)t + \frac{1}{2} \cos(\omega_c - \omega_m)t$$

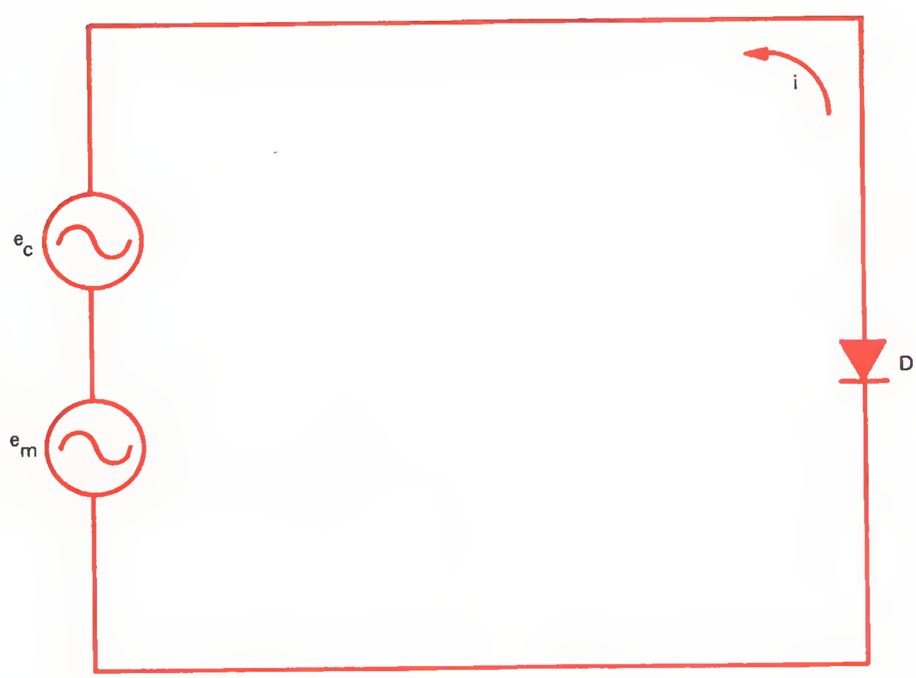


Fig. 10-3 A Nonlinear Mixing Circuit

we will have

	$i = \dfrac{A_2E_c^{\,2}}{2}\cos 2\omega_c t$	Carrier 2nd harmonic
Amplitude Modulated Carrier	$+ A_2E_cE_m \cos(\omega_c + \omega_m)t$	Upper modulation sideband
	$+ A_1E_c \cos \omega_c t$	Carrier fundamental
	$+ A_2E_cE_m \cos(\omega_c - \omega_m)t$	Lower modulation sideband
	$+ \dfrac{A_2E_c^{\,2}}{2}\cos 2\omega_m t$	Modulation 2nd harmonic
	$+ A_1E_m \cos \omega_m t$	Modulation
	$+ \dfrac{A_2}{2}(E_c + E_m)$	Direct current **(10.2)**

This expression is very long indeed and quite different from what we had for linear mixing. Let's look at these seven terms individually. The first is a carrier second-harmonic term. It is considered undesirable and is usually eliminated from the modulated radio signal. This is usually not too difficult since it is at almost twice the frequency of any of the other terms.

The second, third, and fourth terms are the desired amplitude modulation terms. They are called the *upper modulation sideband,* the *carrier,* and the *lower modulation sideband* respectively. Notice that if the carrier is at 1 MHz and the modulation is at 1 kHz, then the *USB* (upper sideband) is at 1.001 MHz while the *LSB* (lower sideband) is at 0.999 MHz. In other words *the three amplitude modulation components are very close together* in frequency.

The fifth, sixth, and seventh terms are all very low-frequency terms. They will not normally radiate and therefore are considered undesirable as far as the modulation process is concerned. However, since they are far removed in frequency from the desired signal they can easily be suppressed.

If we use a tuned circuit to couple the signal out of our simple diode modulator as shown in figure 10-4 we can remove the undesirable signal terms. The tank circuit is

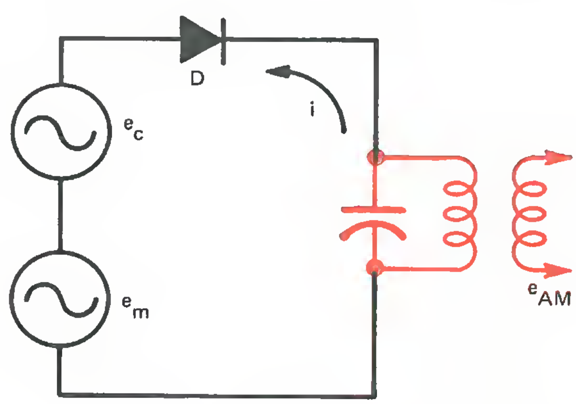


Fig. 10-4 Coupling Out the AM Signal

tuned to the carrier frequency and has a bandwidth broad enough to pass both the USB and the LSB. The bandwidth, however, is much too narrow to pass the undesirable high- and low-frequency components.

The output of this circuit (e_{AM}) will have a carrier component which we will call $E_o \cos \omega_c t$ for simplicity. However, you

should note that E_o in this case will not normally be equal to the original carrier amplitude. Similarly there will be both a USB and a LSB term which we will assign amplitudes $\frac{1}{2} E_s$ which are not normally equal to the original modulation amplitude. The complete amplitude modulated output signal will then have the form

$$e_{AM} = E_o \cos \omega_c t + \frac{1}{2} E_s \cos (\omega_c + \omega_m)t + \frac{1}{2} E_s \cos (\omega_c - \omega_m)t \qquad (10.3)$$

You should compare this expression to the one for linear mixing (equation 10.1). Notice that both have a carrier frequency term, but that a modulated signal has USB and LSB terms that are not produced by linear mixing. All three amplitude modulation terms will radiate, whereas only the carrier term will radiate in the *mixed* case.

A modulated signal looks different than a mixed one on an oscilloscope. Figure 10-5 shows the waveform usually associated with amplitude modulation.

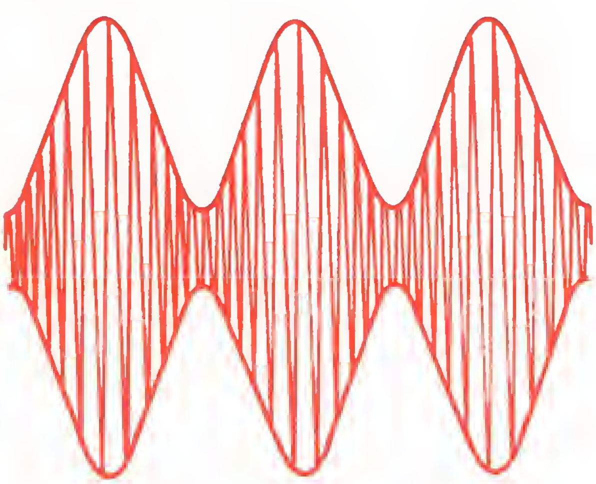


Fig. 10-5 An Amplitude Modulated Signal

It is interesting to notice that we can produce this waveform in two different ways. As we have seen, it can be produced by combining two signals nonlinearly. On the other hand, we can also produce it by *linearly* combining three signals; the carrier, USB, and LSB. Equation 10.2 verifies this second possibility.

One point of confusion that can arise for the inexperienced person develops out of the fact that many cheap AM signal generators do not effectively filter out all of the undesired terms in equation 10.2. As a result it is not too uncommon to encounter a signal generator waveform like figure 10-6. Here we have

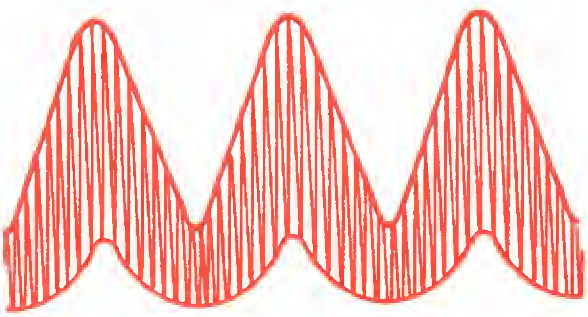


Fig. 10-6 Combined AM and Linear Mixing

all of the terms in equation 10.2 except the first and last one. This waveform results when we *mix* an amplitude modulated signal with the low frequency terms. While such mixing doesn't really interfere with the signal generator's usefulness, it does produce confusing waveforms.

There are several things we can determine from an amplitude modulated waveform. For example, we can measure both its maximum peak-to-peak amplitude (E_{max}) and its minimum peak-to-peak amplitude (E_{min}) as shown in figure 10-7. If we examine equation 10.3 we see that the maximum possible peak-to-peak value that the waveform

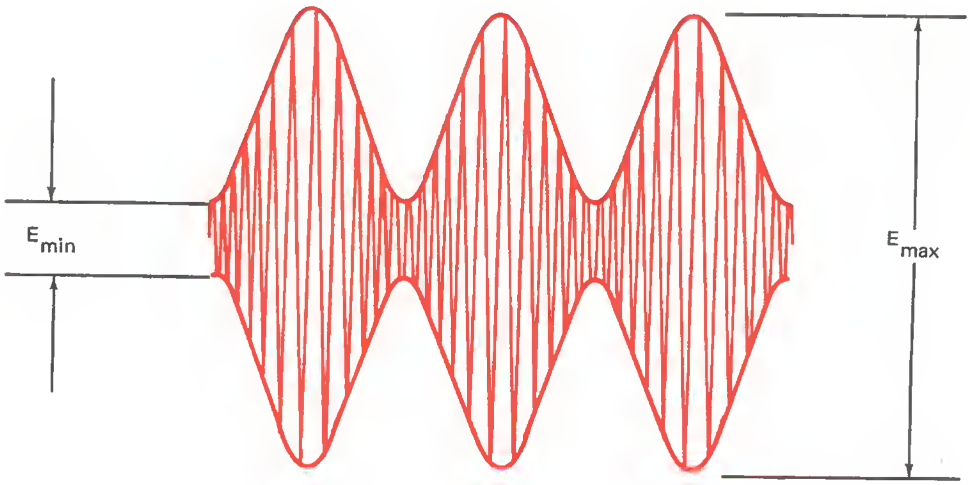


Fig. 10-7 Measuring E_{max} and E_{min}

can have is twice the peak carrier amplitude ($2E_o$), plus twice the USB peak value (E_s), plus twice the LSB peak value (E_s). Therefore, we have

$$E_{max} = 2E_o + E_s + E_s$$

or

$$E_{max} = 2E_o + 2E_s \tag{10.4}$$

Similarly the minimum possible value the waveform can have is twice the carrier amplitude, minus twice the USB amplitude, minus twice the LSB value. That is

$$E_{min} = 2E_o - 2E_s \tag{10.5}$$

Adding these two equations gives us

$$E_{max} + E_{min} = 4E_o$$

or

$$E_o = \frac{1}{4}(E_{max} + E_{min}) \tag{10.6}$$

Then subtracting equation 10.5 from equation 10.4 we have

$$E_{max} - E_{min} = 4E_s$$

or

$$E_s = \frac{1}{4}(E_{max} - E_{min}) \tag{10.7}$$

In other words, measuring E_{max} and E_{min} allows us to evaluate both the carrier amplitude and the sideband amplitude.

The amplitude modulated waveform can also be used to measure the extent to which the carrier is modulated.

This extent of modulation is usually expressed as a *percentage of modulation* and is defined as

$$\boxed{\%\ \text{mod.} = \frac{E_s}{E_o} \times 100} \tag{10.8}$$

Substituting the waveform values for E_s and E_o we get

$$\boxed{\%\ \text{mod.} = \frac{E_{max} - E_{min}}{E_{max} + E_{min}} \times 100} \tag{10.9}$$

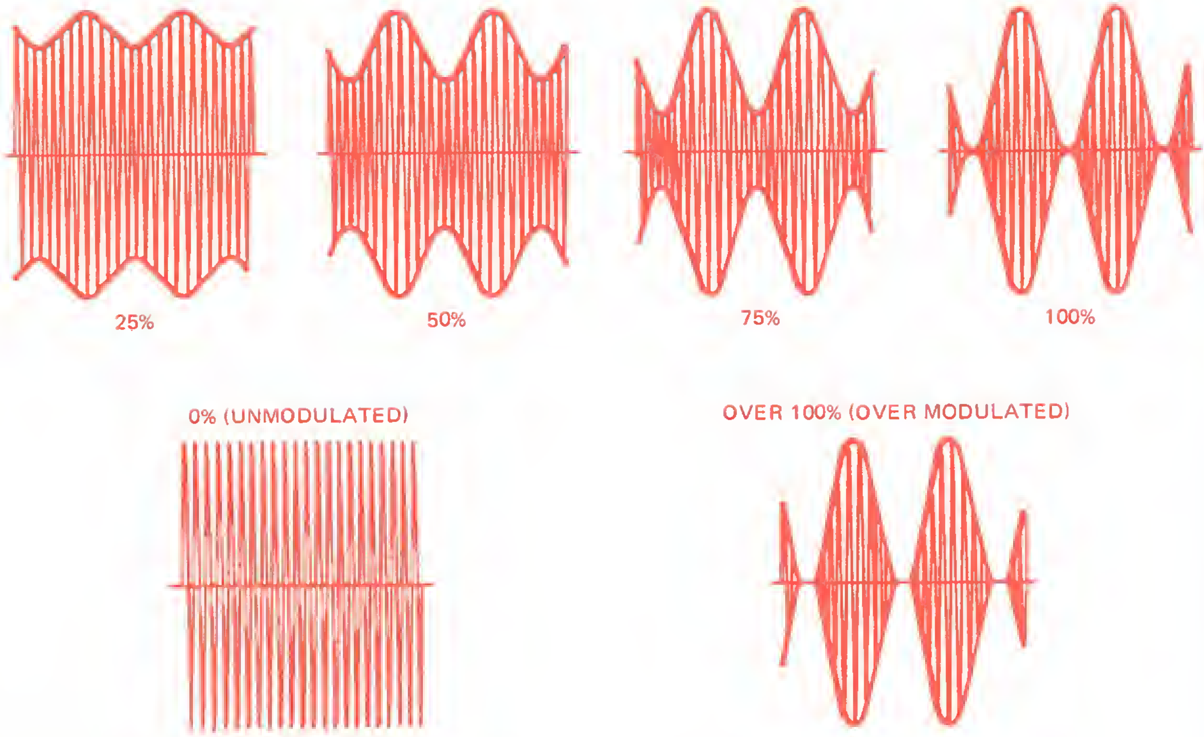


Fig. 10-8 Waveforms for Various Percentages of Modulation

Figure 10-8 shows waveforms for several different percentages of modulation. The carrier amplitudes (E_0) are all approximately the same. Notice that both E_{max} and E_{min} change as the percent of modulation changes. However, the change in E_{min} is considerably more apparent to the viewer.

In the overmodulated case, the modulation envelope is severely distorted. This condition is to be avoided in virtually all radio communications (pulse modulation being a notable exception).

So far we have considered only a diode modulation circuit. In actual practice diodes are rarely used as modulators. It is worthwhile considering them however, since they make it easier to understand that modulation, detection, heterodyning and even rectification are all very similar processes.

It would be considerably more practical to use a transistor for the modulation process. Figure 10-9 shows a simple transistor modulator stage. In this circuit the base-emitter junction is used to perform the modulation. The carrier is coupled into the base through capacitor C_1 while the modulation signal is coupled in through transformer T_1. The collector tank T_2 is tuned to the carrier frequency and is used to filter out the undesirable modulation components as well as to couple the AM signal out of the stage.

The modulation signal can also be coupled into the emitter circuit as shown in figure 10-10. In this case not only is the base current varied by the modulation signal, but so is the collector current.

Probably the most common type of modulator circuit is the collector-modulated stage. While the collector characteristics are not the same as those of a diode, they *are* nonlinear and can be used effectively for

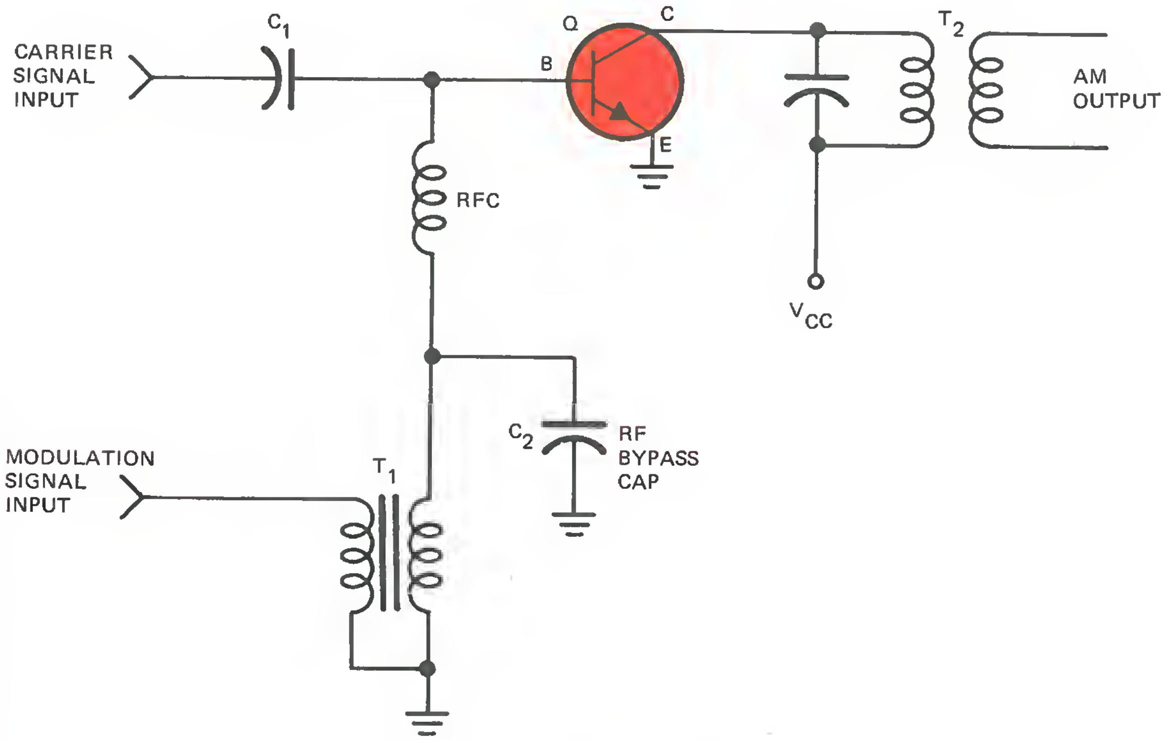


Fig. 10-9 A Base-Modulated Transistor Modulator

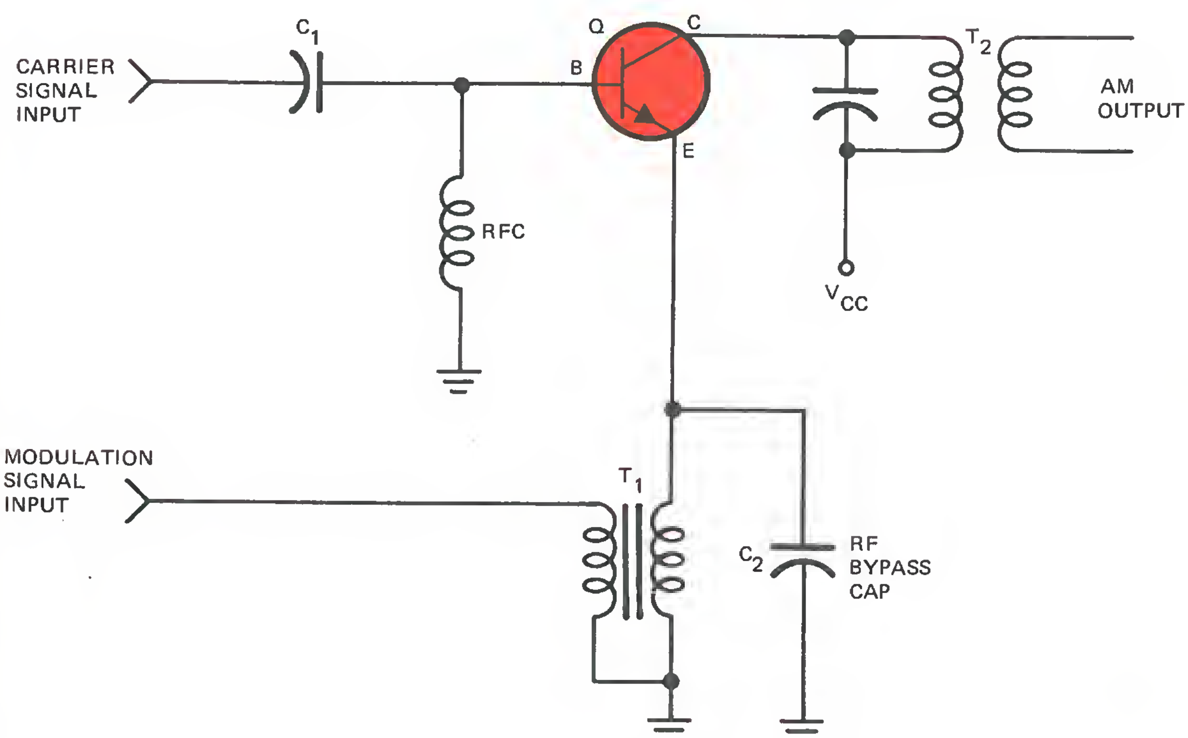


Fig. 10-10 An Emitter-Modulated Transistor Modulator

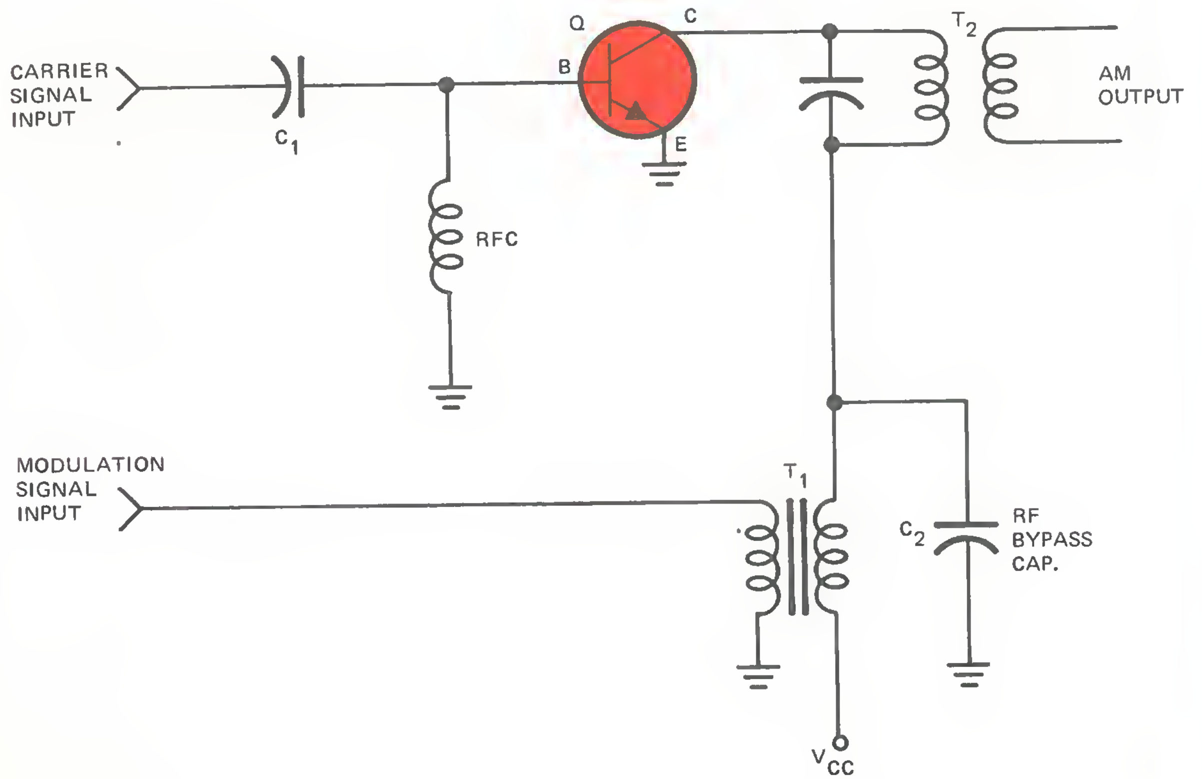


Fig. 10-11 A Collector-Modulated Transistor Modulator

modulation. Figure 10-11 shows a typical collector-modulated stage. In some cases the modulation transformer T_1 is replaced with a class A transistor.

Vacuum tube modulators provide substantially the same types of circuits. That is, they may be control-grid modulated, cathode modulated, or plate modulated. In addition, pentode tubes offer the possibility of screen-grid modulation and suppressor-grid modulation.

Any stage in an AM transmitter can be modulated. However, it is most common to modulate the final power amplifier stage. Such a system is called *high level modulation* and requires substantial amounts of modulating power. Modulating a stage prior to the final power amplifier is called *low level modulation* and requires much less modulating power. However, low level modulation often produces lower levels of RF power output and is for this reason less popular.

In this experiment we will use an AM transmitter of the "wireless microphone" type. The circuit diagram of this transmitter is shown in figure 10-12. Notice that V_2 is an RF oscillator. V_1 is an audio amplifier with its output coupled to the screen grid of the RF oscillator.

This transmitter employs screen-grid modulation applied directly to the oscillator tube. The principles of modulation in this case are the same as those discussed previously.

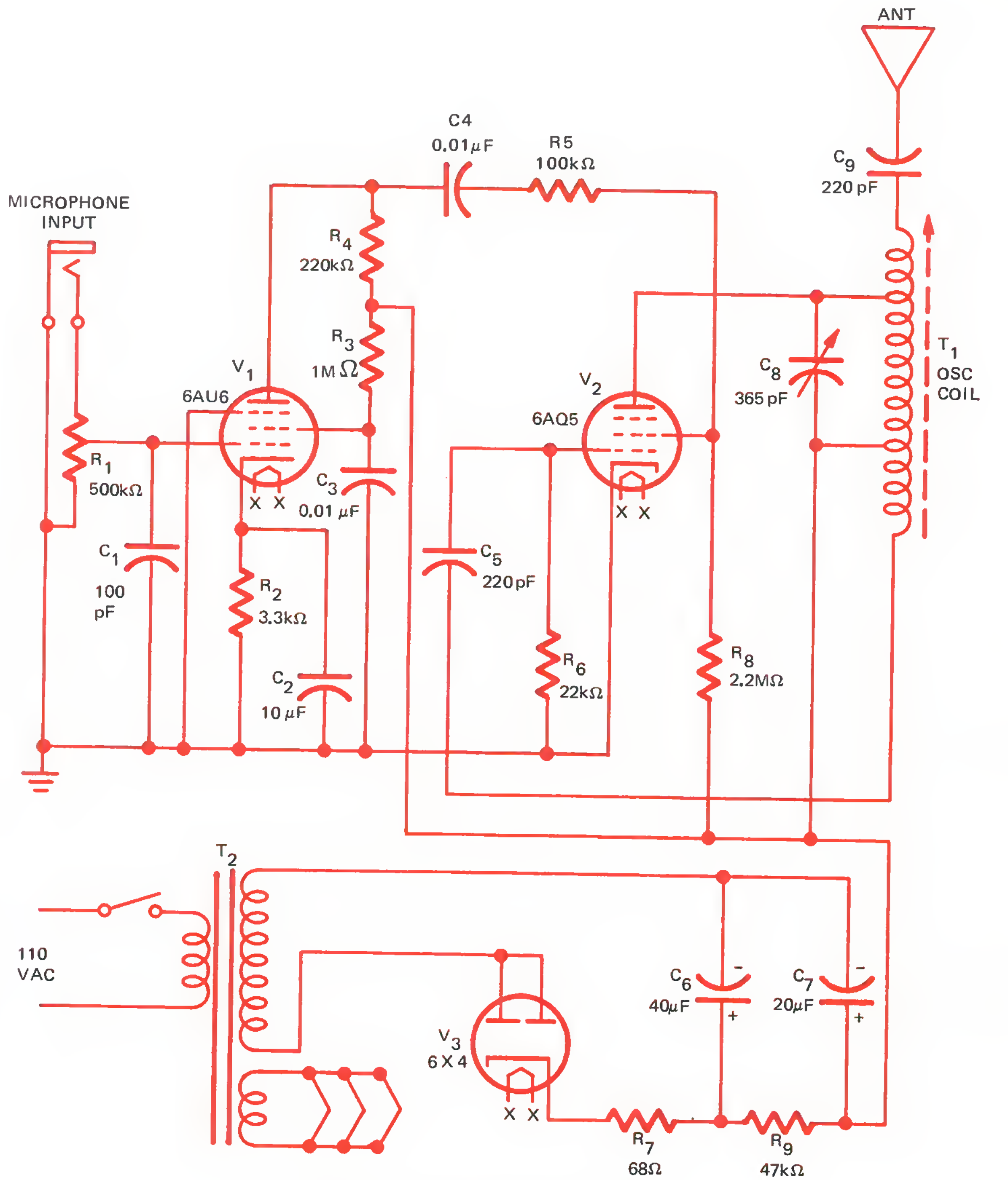


Fig. 10-12 The AM Transmitter

MATERIALS

1 AM radio transmitter
1 Circuit diagram for the transmitter
1 RF signal generator
1 Oscilloscope

PROCEDURE

1. Turn on all of your equipment and allow it several minutes to warm up.
2. Connect the signal generator lead wire to the audio output and set the audio level to zero.
3. Connect the "hot" leads of the signal generator and the oscilloscope together. Do not connect the ground leads at all. You should be able to view a 60-Hz sinewave on the scope. If not, try reversing the AC plug of the signal generator.
4. Now turn up the signal generator audio level until the waveform is about twice the peak-to-peak amplitude of the 60-Hz sinewave viewed in step 3.
5. Sketch the waveform. Label it linear mixing, modulation, or combined mixing and modulation whichever is appropriate.
6. Unplug the microphone from the AM transmitter and connect the audio signal across the volume control.
7. Connect the oscilloscope across the output tank circuit. Use a low capacity or properly compensated x10 probe.
8. Set the signal generator audio level to zero and the transmitter volume control to mid-range. Measure and record the *peak* amplitude (E_o) and the frequency (f_c) of the un-modulated waveform. Sketch the waveform.
9. Increase the audio level until the waveform shows an E_{min} approximately 2/3 of E_{max} (the height observed in step 8).
10. Measure and record; the maximum peak-to-peak amplitude (E_{max}), the minimum peak-to-peak amplitude (E_{min}), the modulation frequency (f_m), and the carrier frequency (f_c). Then sketch the waveform.
11. From the values measured in step 10 compute; the carrier amplitude (E_o), the sideband amplitude (E_s), the upper and lower sideband frequencies (f_{USB} and f_{LSB}).
12. Increase the audio level until the waveform has an E_{min} approximately 1/2 of E_{max}. Then repeat steps 10 and 11.
13. Again increase the audio level until E_{min} is approximately 1/3 of E_{max}. Then repeat steps 10 and 11.
14. Increase the audio level (and the volume control setting if necessary) until E_{min} is almost zero. Then repeat steps 10 and 11.
15. Increase the audio level (and volume control setting if necessary) until the waveform is severely over-modulated. Sketch the waveform.
16. Disconnect the scope and signal generator from the transmitter. Connect the signal generator lead wire to its RF output. Then connect the generator to the scope so that you can view the generator output.

Waveform from step 5

Type of waveform = ____________________

Step 8

Step 10

E_o = ____________
f_c = ____________

E_{max} = __________ f_c = __________
E_{min} = __________ f_m = __________
E_o = __________ f_{USB} = __________
E_s = __________ f_{LSB} = __________

Step 12

Step 13

E_{max} = __________ f_c = __________
E_{min} = __________ f_m = __________
E_o = __________ f_{USB} = __________
E_s = __________ f_{LSB} = __________

E_{max} = __________ f_c = __________
E_{min} = __________ f_m = __________
E_o = __________ f_{USB} = __________
E_s = __________ f_{LSB} = __________

Fig. 10-13 The Data Tables

Step 14

Step 15

E_{max} = ______ f_c = ______

E_{min} = ______ f_m = ______

E_o = ______ f_{USB} = ______

E_s = ______ f_{LSB} = ______

The overmodulated waveform

Signal Generator Waveforms

1

f = ______ type = ______

2

f = ______ type = ______

3

f = ______ type = ______

4

f = ______ type = ______

5

f = ______ type = ______

6

f = ______ type = ______

Fig. 10-13 The Data Tables (cont'd.)

17. Set the generator controls for maximum modulated output at its lowest RF frequency. Sketch the modulated RF output waveform and record the frequency.

18. Switch the generator to its next higher frequency *range* and sketch the waveform. Do this for each of the frequency ranges of the generator. (Use the minimum frequency within each range and record its value each time.)

19. Label each of the waveforms taken in steps 17 and 18 as: mixing, modulation, or combined mixing and modulation as the case may be.

ANALYSIS GUIDE. In the analysis of these data you should consider the extent to which your results agreed with the material presented in the discussion. In particular consider whether the value of E_o changed from step 8 through 14. Explain in your own words why this occurred.

PROBLEMS

1. Sketch a waveform of two mixed sinewaves and explain how you would measure the amplitude and frequency of each sinewave.
2. Explain why you can't calculate percentages of modulation above 100% using the method in the discussion.
3. What is the relationship between the sideband frequencies and the frequencies of the carrier and modulation signals?
4. Explain how a 400-Hz signal would sound in a receiver if the percent of modulation of the incoming signal were:

   (a) 25%
   (b) 50%
   (c) 75%
   (d) 100%
   (e) over 100%

5. What is the difference between mixing and modulation?
6. How can you tell when both modulation and mixing are present in a waveform?

experiment 11 PULSE TELEMETRY SYSTEMS

INTRODUCTION. Telemetry can be considered to be any transmission system that is used primarily for remote processing of data. In this experiment we shall examine some of the more commonly used types of telemetry systems.

DISCUSSION. The word telemetry actually means measuring from a distance. Telemetry is not usually the transmission of voice messages or music such as in radio broadcasting or two-way radio communications systems. It involves measuring some quantity, conversion of this quantity into a coded electrical signal, and then transmitting this signal to its destination. At the destination the signal is received and decoded into some usable form for observation and measurement. The measurement of the heartbeat of an astronaut, the speed of a missile or the temperature of a remotely located storage tank are all typical quantities for telemetry transmissions. The results of these measurements may be transmitted over wires such as telephone lines, laser beams, microwave beams, or by a wireless radio system. The information thus received is commonly read out or displayed on a meter, or oscilloscope. It is often recorded on a chart recorder or magnetic tape for later use and verification.

Remote control can be considered the reverse of the telemetry process in that a quantity is measured and then sent to a remote location to perform a given operation such as in the guidance of a drone aircraft or missile.

The data we wish to transmit can be converted into pulses of various shapes and then transmitted to the desired location. This conversion of data into pulses can be done in a variety of ways and is called encoding. Three of the most common types of pulse telemetry or remote control systems are:

PAM or *pulse amplitude modulation* in which the pulse height is proportional to the data value.

PDM or *pulse duration modulation* in which the pulse duration (width) is proportional to the data value.

PPM or *pulse position modulation* in which the position of a pulse with respect to a reference pulse is proportional to the data value.

In all of these systems the data that is being measured is sampled at uniform intervals and the data value at the instant of sampling is converted to the desired pulse parameter. To better see this action let us look at a varying data value and compare the results of the three types of conversions into pulse waveforms as shown in figure 11-1.

Depending on the situation these data values could be sampled once each millisecond, once each second, or even once a day.

Now let's look at a method by which we can perform PAM encoding. The parameter to be measured is converted into an electrical signal by means of a transducer. This signal is then sampled for a given period of time at a given rate. For instance, suppose we wanted to monitor the skin temperature of a missile during a test flight. We could attach a ther-

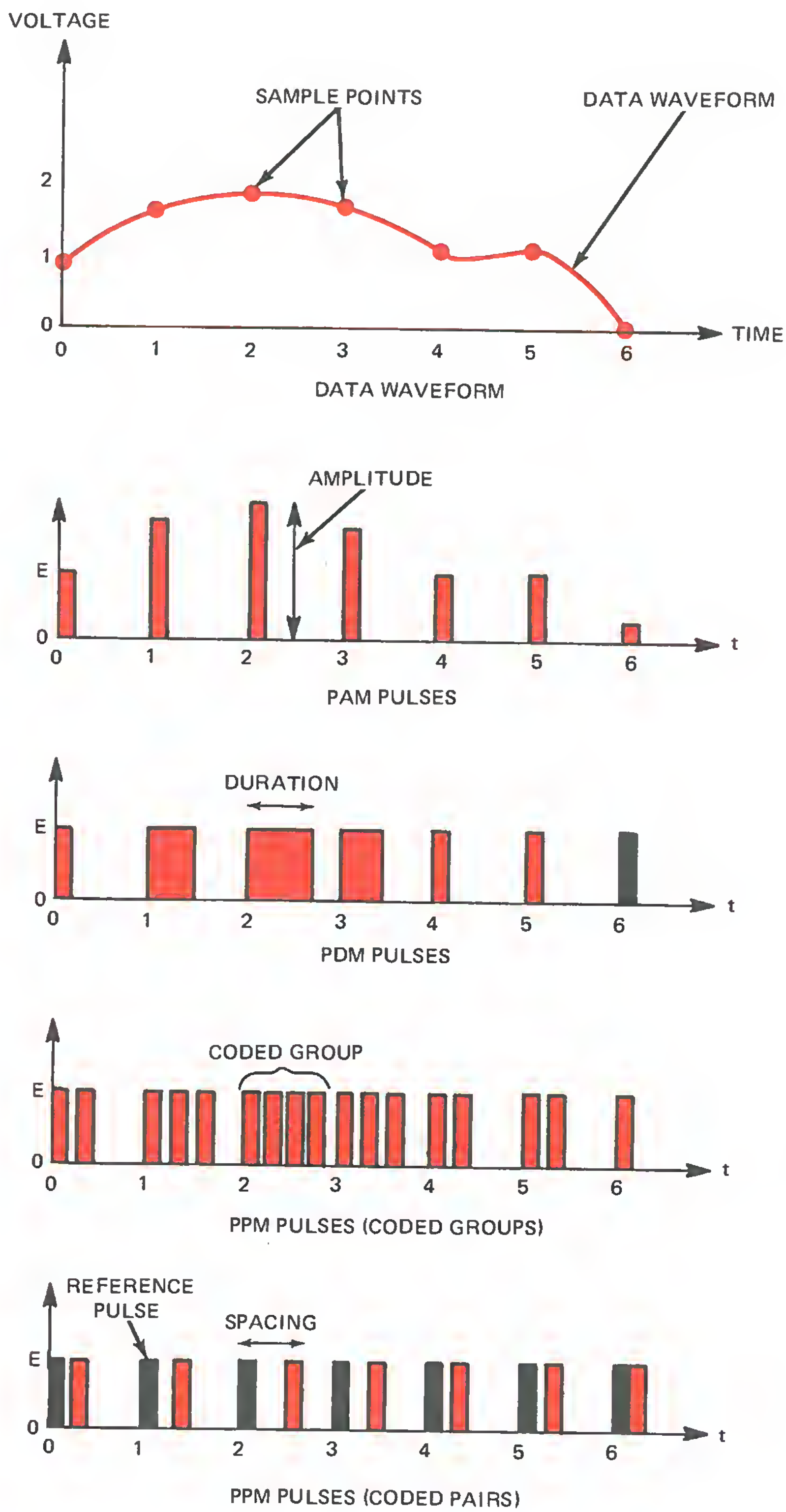


Fig. 11-1 Common Types of Pulse Telemetry

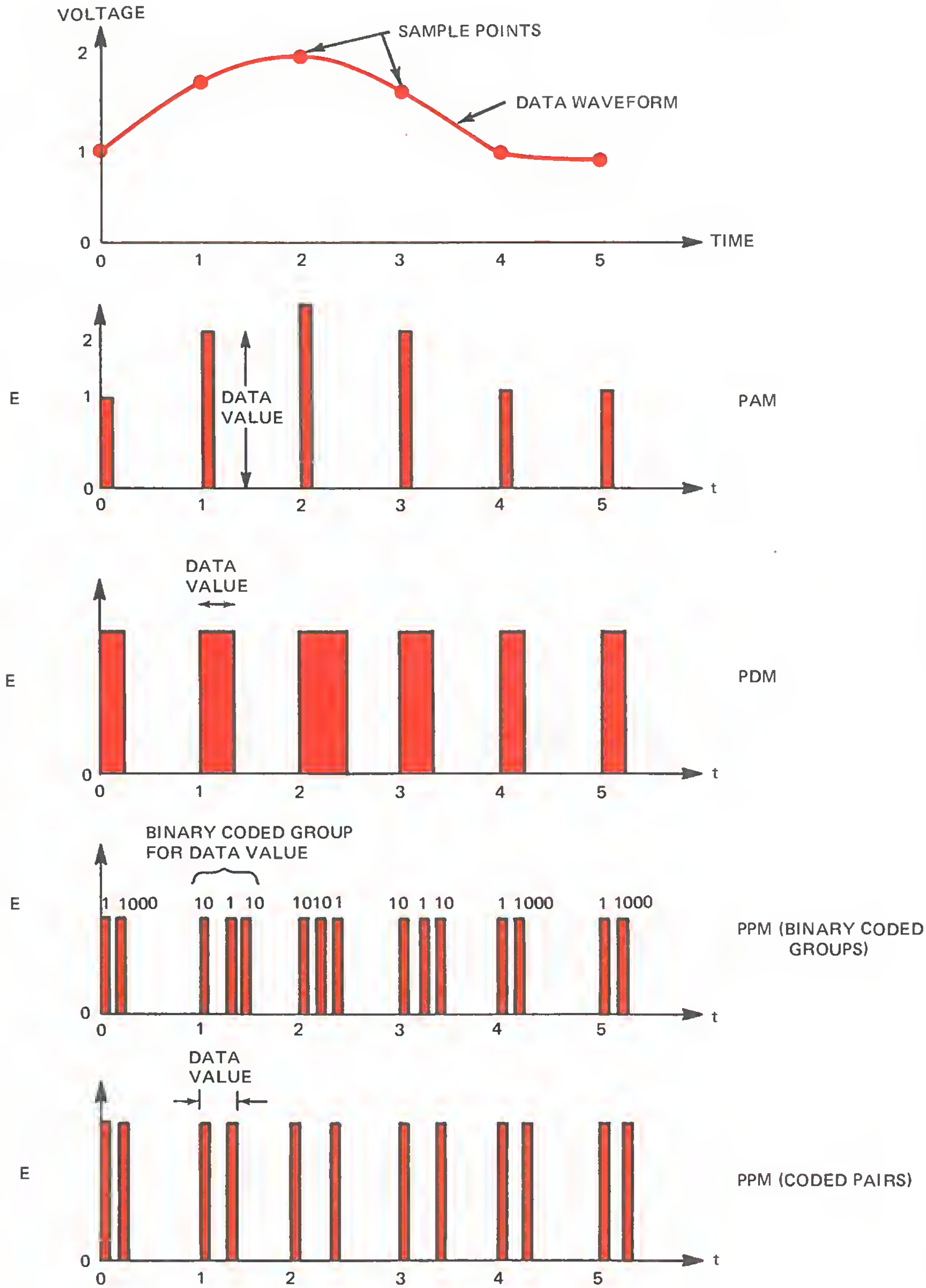


Fig. 11-2 Pulse Encoding

mistor to the skin of the missile. This thermistor would be our transducer converting the skin temperature to a corresponding resistance value. The thermistor could then be part of a voltage divider so our resistance value would now be proportional to a voltage level as shown in figure 11-3.

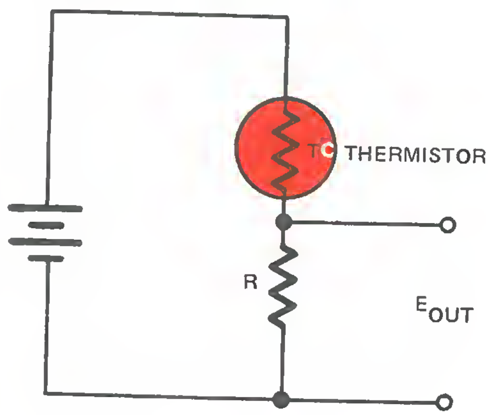


Fig. 11-3 Thermistor Voltage Divider

The thermistor might have a negative temperature coefficient and thus decrease in resistance value as the temperature increased.

This reduction in resistance would increase the current flow in the circuit causing the voltage across R to increase. This relationship can be presented on a chart as shown in figure 11-4.

This relationship can also be described mathematically

$$E_R = \frac{T}{20} - 1.25 \qquad (11.1)$$

For example, when T = 75°F(point A)

$$E_R = \frac{75}{20} - 1.25 = 3.75 - 1.25 = 2.5V.$$

Let us proceed further and add a switch to the circuit so we can sample this voltage for a given period of time. We will call this switch a gate and activate it with a pulse having a duration equal to the period of time we wish to sample this voltage. The circuit will now appear as in figure 11-5.

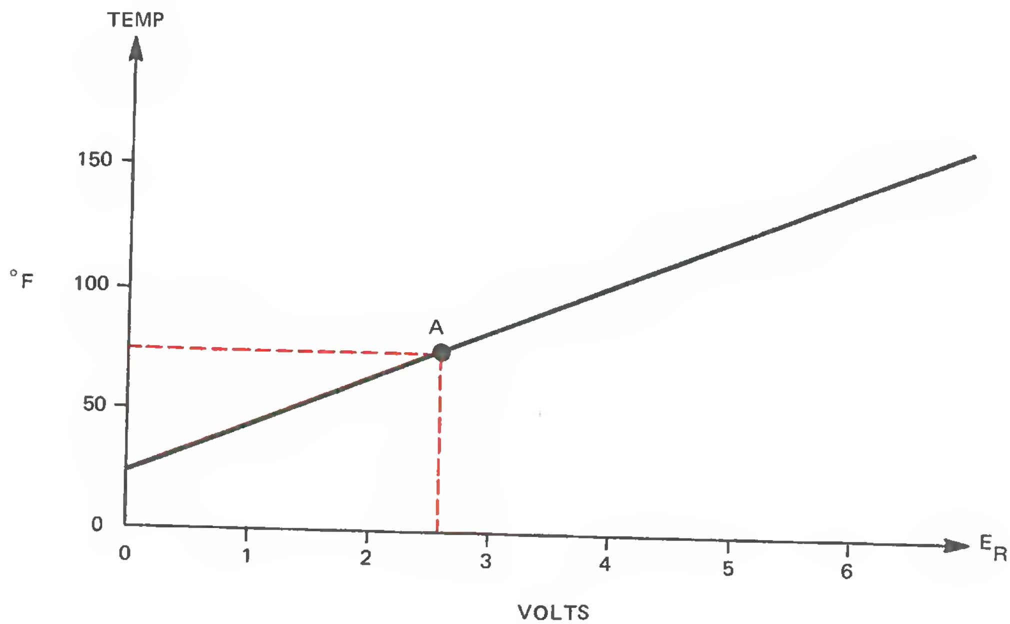


Fig. 11-4 Temperature-Voltage Relationship

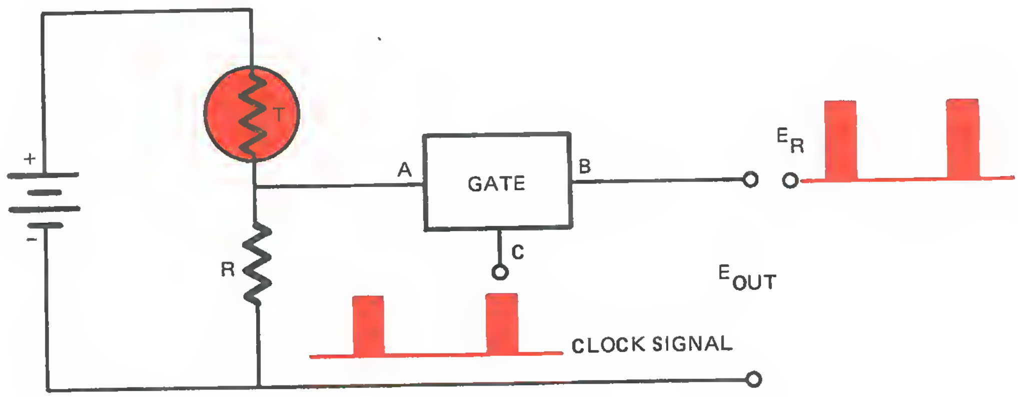


Fig. 11-5 Gated Signal

Now each time a clock pulse is present at C we will have a pulse appear at the output terminals of amplitude E_R. The greatest problem with PAM is in its poor noise rejection. If we are transmitting a PAM signal and we have large amounts of noise (interference) present, the receiver sees the noise as signals and can give false readings as shown in figure 11-6.

A PDM system is not usually as susceptible to noise as a PAM system. The information is contained in the width of the pulses and not in their height. We can develop a PDM signal by triggering a monostable multivibrator at regular intervals called the *clock rate*. A suitable transducer can then control the width of each pulse as shown in figure 11-7.

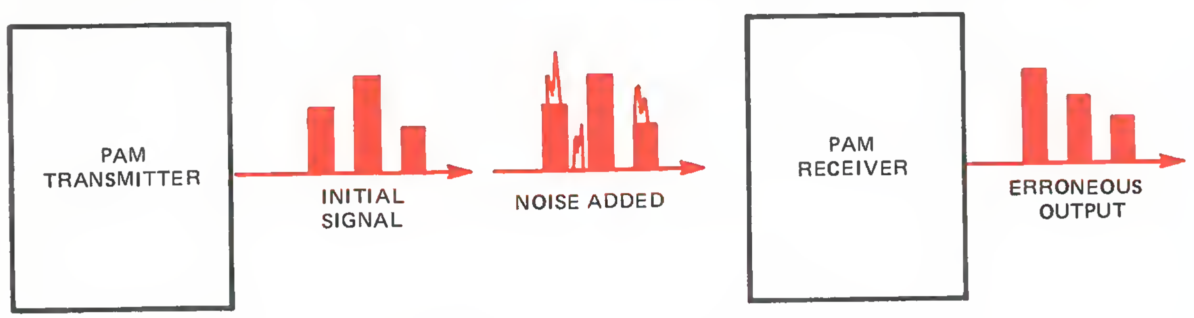


Fig. 11-6 Effect of Noise on a PAM Transmission

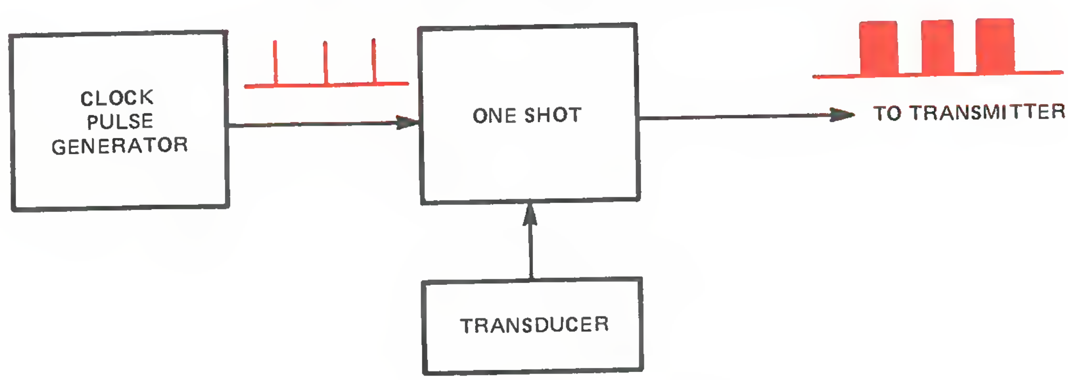


Fig. 11-7 PDM Encoder

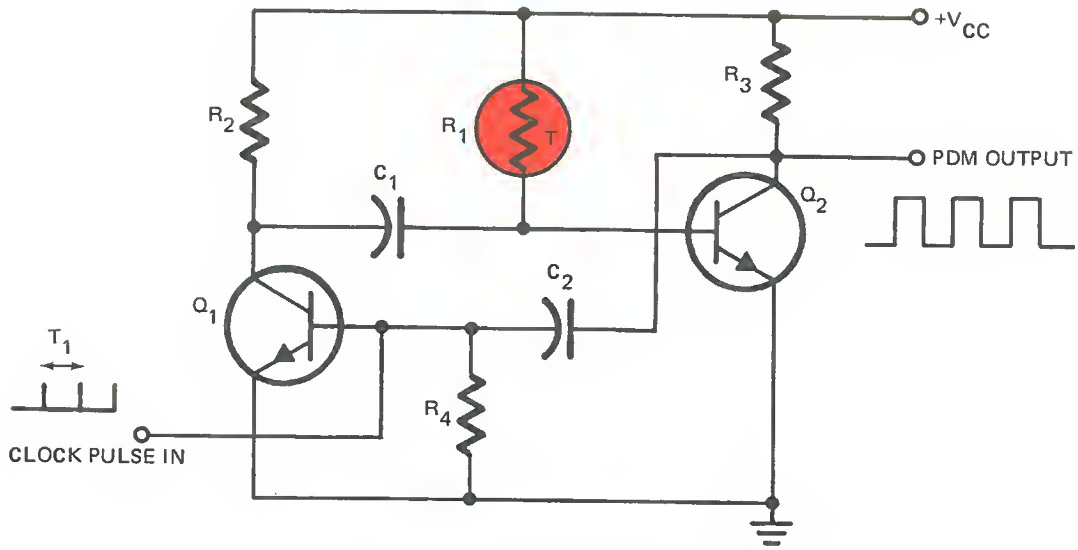


Fig. 11-8 Multivibrator PDM Encoder

Looking at figure 11-8 we can see that Q_1 is biased off by R_4 while Q_2 is biased on by R_1. These transistors will remain in the stable state until a clock pulse at the base of Q_1 turns it on.

When this occurs, the collector of Q_1 will start to become more negative. This negative going voltage change will be coupled thru C_1 and will attempt to turn Q_2 off. This will generate a positive going voltage pulse at the collector of Q_2 which will be coupled thru C_2 to the base of Q_1 driving it into saturation. This regenerative feedback between Q_1 and Q_2 will result in Q_1 being turned fully on and Q_2 being turned fully off.

Going back to our original stable state of Q_1 off and Q_2 on, note that C_1 will be charged with its R_2 side positive in relation to its R_1 side. When the circuit is triggered into its other state, Q_1 will clamp the positive end of C_1 to ground. This will result in the base of Q_2 being driven to a negative potential due to the charge on C_1.

The circuit will now stay in this state until C_1 discharges thru R_1 to a level which allows Q_2 to start changing states again. When Q_2 starts to turn on, the regenerative action will occur and the circuit will return to its stable state until triggered again by another clock pulse.

The time of T_2 is thus determined by R_1C_1 and can be approximated by

$$\boxed{T_2 = 0.69\ R_1C_1} \tag{11.2}$$

For example; if $R_1 = 22\ k\Omega$ and $C_1 = 0.1\ \mu F$, then

$$T_2 = 0.69\ R_1C_1$$

$$T_2 = 0.69(2.2 \times 10^4)(10^{-7})$$

$$T_2 = 1.52 \times 10^{-7} \text{ or } 1.52 \text{ ms}$$

The same thermistor used in the previously described PAM system could be used to control the pulse duration as shown in figure 11-8.

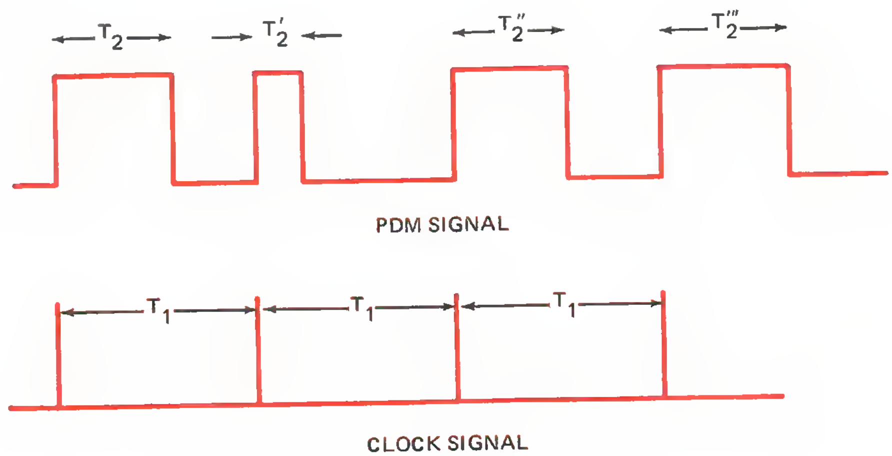


Fig. 11-9 Output of a PDM Encoder

This could give us the resulting waveform shown in figure 11-9. The period T is controlled by the frequency of the clock pulse and remains fixed. However, the duration T_2 is a function of R_1C_1. If R_1 is our thermistor and changed in resistance as the skin temperature of the missile changes, the period T_2 is then actually a function of temperature. We now have a pulse train in which the width of the pulses is proportional to the temperature of R_1.

Another pulse system that lends itself to use with a digital computer is the PPM system employing coded groups. In this system all the pulses are of uniform height and width. The information is transmitted in groups of pulses or *words*. These words are in binary code with the 1 condition represented by a pulse and the 0 condition represented by the absence of a pulse as shown in figure 11-10. A block diagram of this type of encoding system would involve a circuit called an analog-to-digital converter or *A-D converter*. This circuit monitors the transducer and produces a binary-coded word each time it is triggered much like the monostable multivibrator produces a pulse of a specific width each time it is triggered. Figure 11-11 is typical of this type of system.

The final system we shall look at is the PPM system employing coded pulse pairs. Like the coded group system the coded pair system uses pulses of uniform height and width. In this system the information is represented by the

DECIMAL DATA VALUE	BINARY DATA VALUE
5	101
7	111
6	110

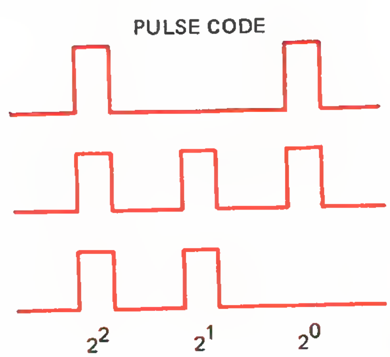


Fig. 11-10 Binary Coding

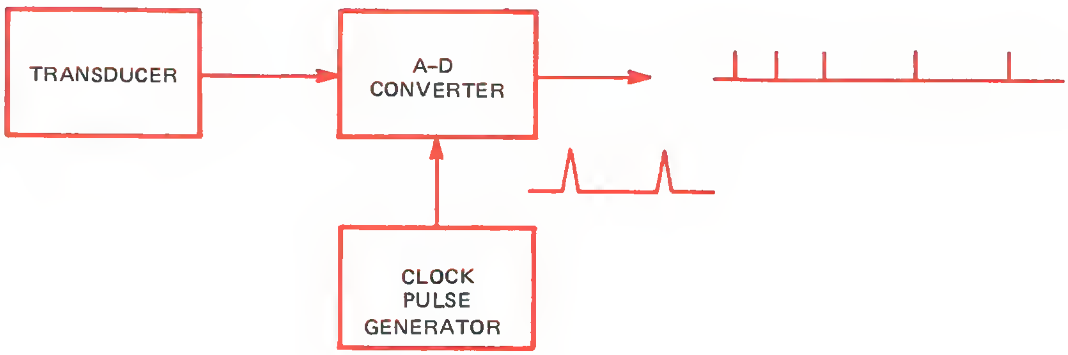
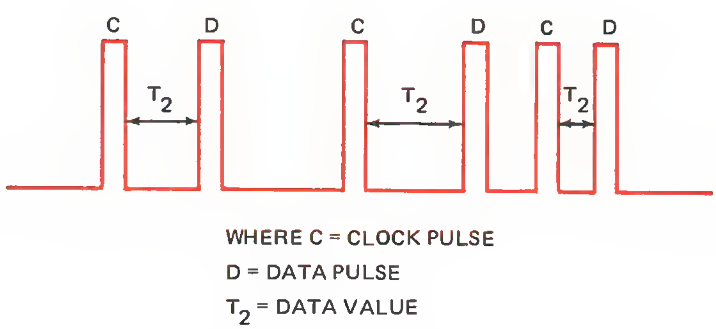


Fig. 11-11 PCM Encoder

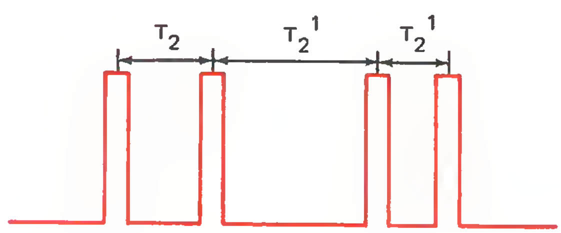
position of a data pulse relative to a reference pulse. The reference pulse could be the transmitted clock pulse. Sometimes the data pulse is referenced to the previous data pulse, in which case we call the system a self-referencing system. Both types of systems are shown in figure 11-12.

The clock-referenced system can be developed by differentiating and fullwave rectifying a PDM pulse train as shown in figure 11-13.

One of the main advantages of PPM is in the power conserved in the transmission of a PPM signal over that of a PDM signal. The



(A) A CLOCK-REFERENCED PULSE TRAIN



(B) A SELF-REFERENCING PULSE TRAIN

Fig. 11-12 PPM Pulse Trains

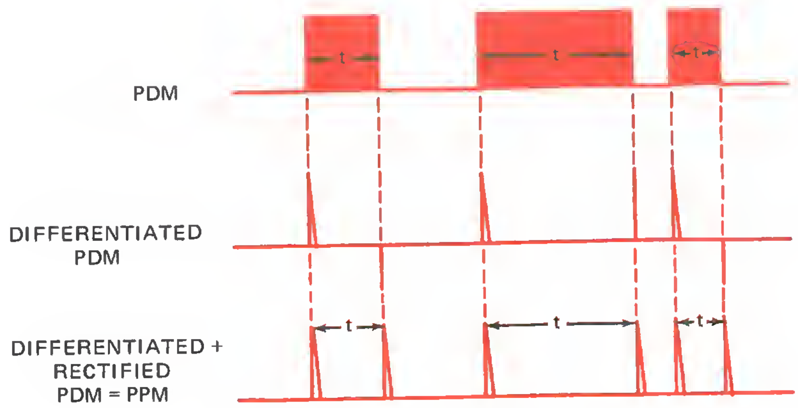


Fig. 11-13 PPM Pulse Train

power required is approximately equal to the area under the transmitted pulses and as indicated in figure 11-13 there is very little area under the PPM pulses in comparison to the PDM pulses, and so less power is required in the PPM system.

The system we will investigate in the laboratory is of the self-referencing type. In this system we will send out a series of six pulses called a frame. This frame is repeated about every 16 ms. The pulses will appear in groups with the first five periods varying from about 1 ms to about 2 ms in length, followed by an off period of from about 6 ms to 11 ms.

By means of this pulse train we can control or monitor five different sets of data. Each set of data is called a channel and is represented by each of the first five periods as shown in figure 11-14. The period of each channel is set by a control on the transmitter and will control the position of an appropriate servo package connected to the receiver. As a transmitter control is moved, the period of its channel will vary from about 1 ms up to about 2 ms. But it will not affect any other channel. In addition, some of the controls will have a fine positioning or *trim* control as well.

The position of each servo can be set independently over a range of about 100 degrees of rotation and each channel is sampled each 16 ms so in effect all the channels can be changed simultaneously and all the servos will respond simultaneously. This sampling of multiple channels in succession is known as multiplexing.

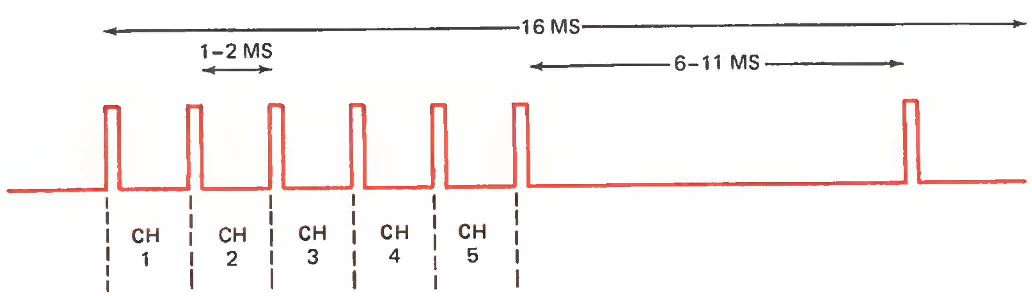


Fig. 11-14 Digital Pulse Train

MATERIALS

- 1 Radio control transmitter, Heath Co. GDA-19-1 or equivalent
- 1 Radio control receiver, Heath Co. GDA-19-2 or equivalent
- 1 Servo, Heath Co. GDA-19-4 or equivalent
- 1 Battery pack and wiring harness, Heath Co. GDA-19-3 or equivalent
- 1 Oscilloscope
- 1 Disk dial

PROCEDURE

1. Make sure all power switches on the radio control system are turned off.
2. Connect the receiver and servo package as shown in figure 11-15. Pay particularly close attention to see that the color code on the wiring harness matches the color code on the master plug.

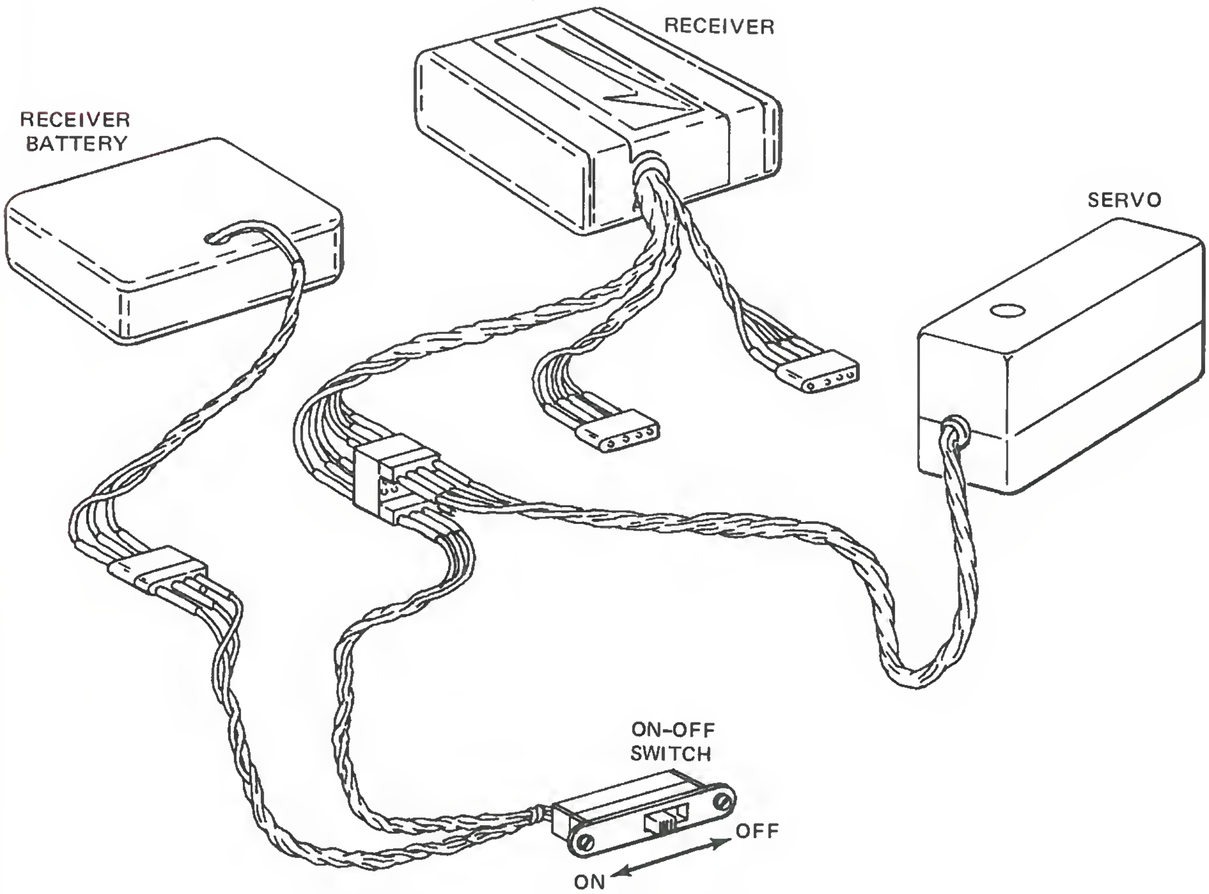


Fig. 11-15 The Experimental Set Up

3. Attach a disk dial and indicator to the rotary servo output wheel as shown in figure 11-16. The disk dial can be attached with tape.

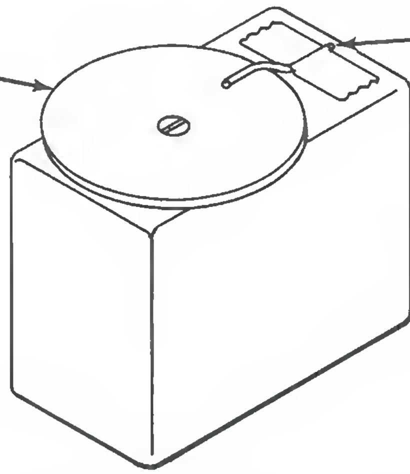


Fig. 11-16 Attaching the Dial and Indicator

4. When applying power to the system always follow this sequence:
   A. Turn on the transmitter before turning on the receiver.
   B. Turn off the receiver before turning off the transmitter.
   NOTE: This will avoid causing damage to the servos due to reception of stray signals.

5. Leave the receiver antenna coiled up on the receiver case and place the transmitter approximately three feet away with its antenna in the retracted position.

6. Turn ON the transmitter main power switch and note if the meter on the transmitter indicates it is operating. It will not reach full scale with the antenna retracted because of improper loading.

7. Turn the receiver power switch ON and note the action of the servo. The servo should shift position slightly and hold. If the servo runs to its limit, in either direction, immediately turn OFF *both* power switches in correct order and notify instructor.

8. Prepare a sketch of the transmitter control layout and the receiver plugs.

9. Connect the servo to each of the channels of the receiver plug in succession. *Make sure the receiver is turned off during channel changes.* By moving the transmitter controls determine which channel on the receiver is associated with which control on the transmitter. Indicate this relationship on the sketch prepared in step 8. Read step 10 before performing this action.

10. Record the range of servo travel in degrees for each full range movement of the transmitter control. Record the range of each trim lever where applicable in the data table.

11. Connect the oscilloscope probe tip to test point A on the receiver (See figure 11-17). Connect the oscilloscope ground lead to the negative battery lead (black wire). Accurately record the signal at this point. This is the signal developed by the receiver before it is decoded. Note and draw the action resulting from moving channel number one and channel number four control sticks through their full range.

(CIRCLES REPRESENT TEST POINTS)

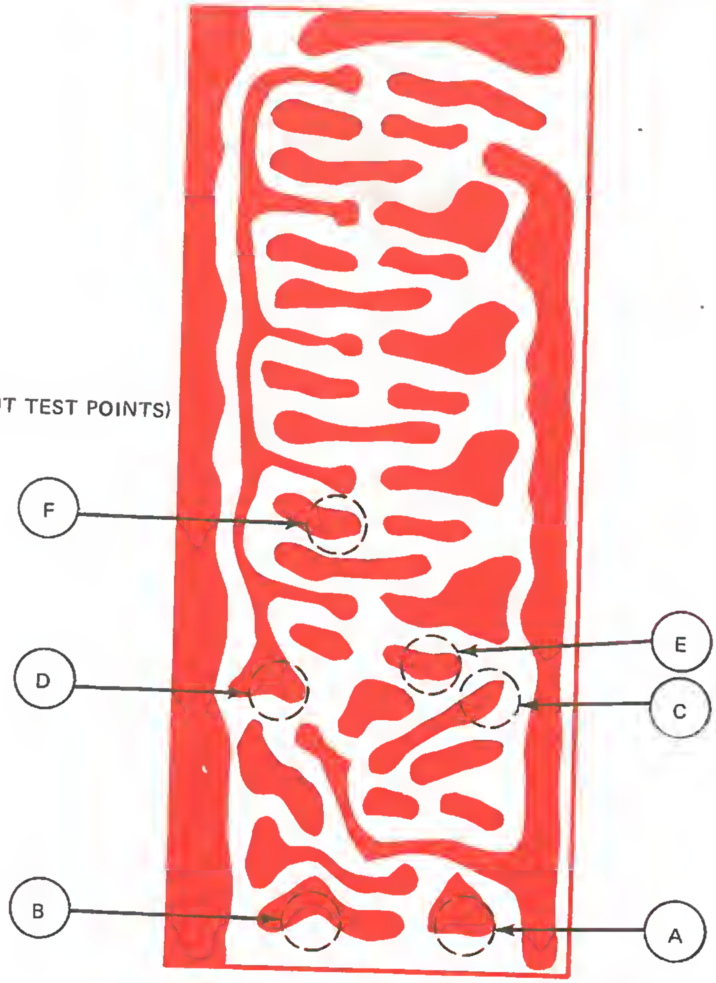


Fig. 11-17 Foil Side of the Receiver

ANALYSIS GUIDE. In analyzing these results you should explain in your own words how each of the five channels can work independently of the others. Why doesn't one of the setups in the lab interfere with the others?

What do you think the trim controls on the transmitter would be used for?

PROBLEMS

1. What is the frequency of the transmitted pulse train?
2. In which direction does the servo rotate as the channel period is shortened?
3. Referring to equation 11.1 what voltage would be developed at T = 122°F?
4. If R_1 = 37 kΩ and C_1 = 0.27 μF, what would be the on period of the one shot multivibrator in figure 11-8?

5. What is the function of R_4 in figure 11-8?

6. Which is more susceptible to interference from noise, PAM or PDM?

7. What is the greatest advantage of PPM over PDM?

8. List four possible uses of telemetry systems.

Fig. 11-18 The Data Tables

Channel No.	Main Control Range	Trim Range
1		
2		
3		
4		
5		

Fig. 11-18 The Data Tables (cont'd.)

experiment 12 DIGITAL ENCODER CIRCUITRY

INTRODUCTION. Encoding is the process of converting data values into a desired code form for signal processing. In this experiment we will examine one method of encoding a PPM pulse train.

DISCUSSION. Let's assume that we want to have a system that will enable us to control five separate noninteracting control functions. We plan to do this by digital or pulse means, using a pulse train like the one shown in figure 12-1. This train is composed of six pulses which are repeated every 16 ms in a group which we will call a frame. Each pulse is of the same amplitude and about 350 microseconds wide. All the pulses in a frame except the first one normally start 1.5 ms after the start of the previous pulse.

The time interval between the first pulse in one frame and the first pulse in the next frame is always about 16 ms. The frame rate is thus fixed. However, the time interval between any two successive pulses within a frame can be increased or decreased by 0.5 ms from the nominal 1.5 ms. Thus the time interval between two successive pulses could be anywhere between 1.0 ms to 2.0 ms. It is this variable time interval that is used to carry the data. By using six pulses we can convey five channels of data which are continuously updated each 16 ms as shown in figure 12-2. Notice that when the channel 4 data changes the frame interval stays constant. Also, the intervals of all of the other channels (except channel 4) remain constant too. All five channels could be changed at once with no interaction between channels.

The long space between the last pulse in a frame and the first pulse in the next frame is called the *sync pause.* This pause has an interval always significantly longer than that of any of the other data channels. For instance, the shortest interval the sync pause can have is 6 ms when all five of the data channels are at their maximum interval of 2 ms.

$$\text{Sync Pause} = \text{Frame interval} - \sum_{1}^{5} \text{channel intervals} \qquad (12.1)$$

$$\text{Sync Pause} = 16 \text{ ms} - (2 + 2 + 2 + 2 + 2) \text{ ms}$$

$$\text{Sync Pause} = 16 \text{ ms} - 10 \text{ ms} = 6 \text{ ms}$$

This shortest sync pause interval of 6 ms is still three times as long as the longest chan-

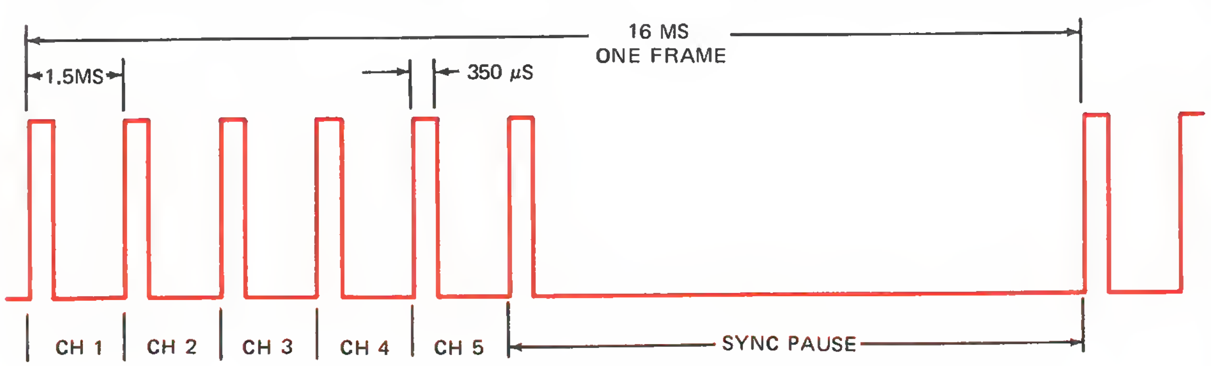


Fig. 12-1 Pulse Train

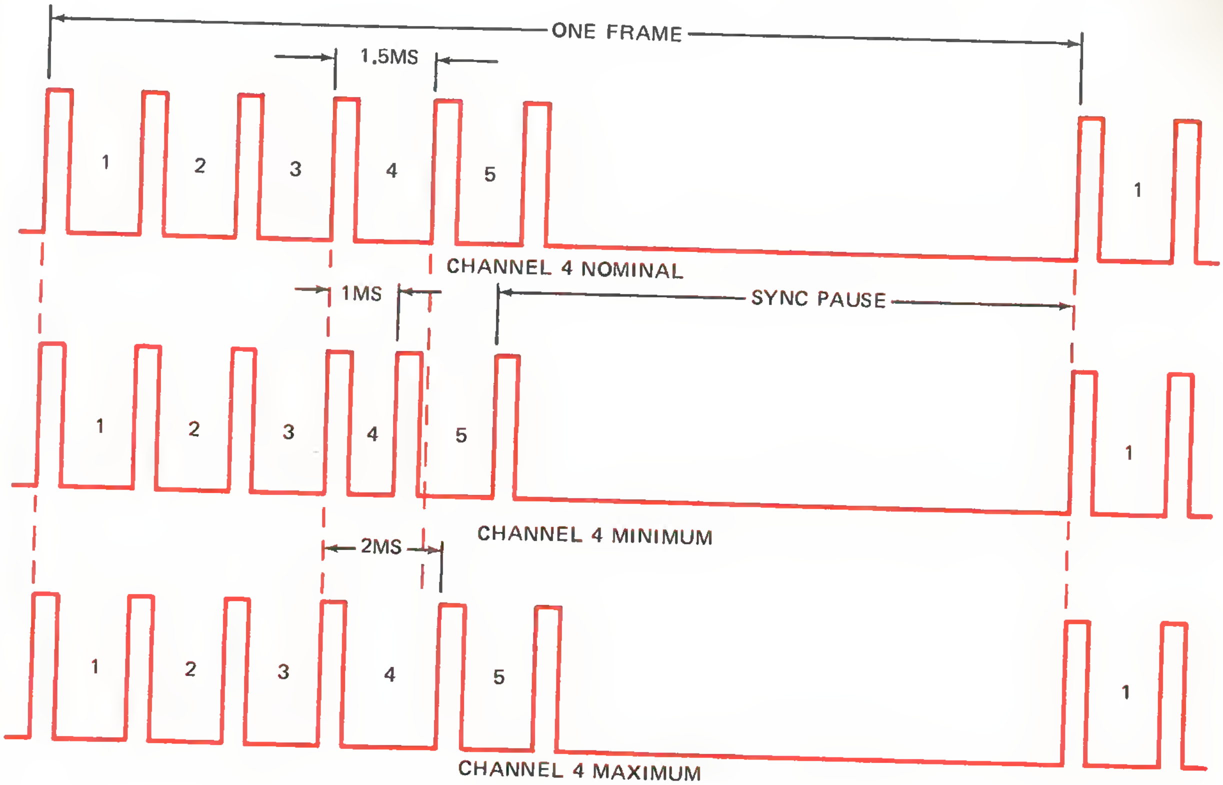


Fig. 12-2 Channel Ranges

nel interval of 2 ms. The decoder that receives this pulse train uses this sync pause to synchronize the encoder and decoder so that their channels coincide. Anytime the decoder detects an interval longer than approximately 4 ms it is reset so that the next incoming pulse will be interpreted as the beginning of the channel one interval. Due to the fact that a sync pause is developed every 16 ms, the system is very tolerant to noise and erroneous data pulses. If a noise pulse is present in the frame and upsets the data interval of a channel, it will only cause this error for 16 ms before a new interval corrects the error. However, if noise is present in such quantity that there is no interval as long as 4 ms, the decoder cannot reset itself and synchronization will be lost. The noise shown in figure 12-3

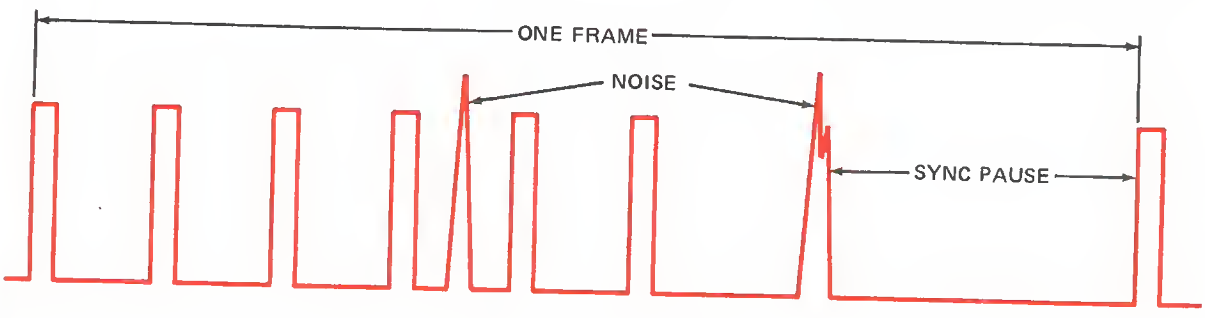


Fig. 12-3 Tolerable Noise

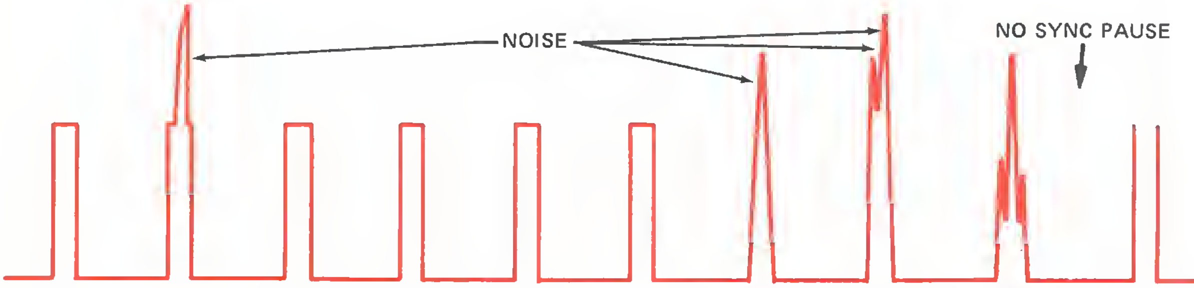


Fig. 12-4 Intolerable Noise

can be tolerated but that shown in figure 12-4 cannot.

Figure 12-3 will cause a temporary error often called a "glitch", while figure 12-4 will cause a complete loss of control if the noise persists. Notice that in figure 12-3 the first three channels will be unaffected by the noise.

Additional noise tolerance is provided by the decoder circuitry in that pulses over a minimum amplitude are detected and treated as part of a pulse train while any pulse amplitude below a selected minimum will be disregarded as shown in figure 12-5.

Thus, if the signal level can be kept well above the noise level and the minimum selected level can also be above the noise level, there will be virtually no interference from noise.

Absolute freedom from interference is almost impossible to achieve and some sort of

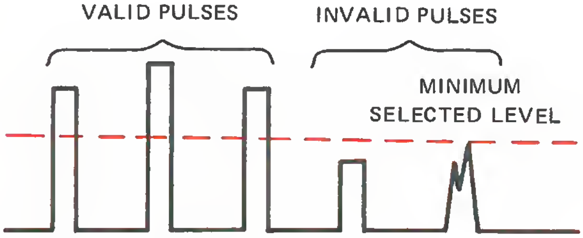


Fig. 12-5 Pulse Threshold

synchronization such as the sync pause is an important part of any digital data transmission system.

Now let's look at the electronic circuitry necessary to produce the previously described pulse train.

A block diagram of the encoder and the RF transmitter it could drive is shown in figure 12-6.

The astable multivibrator operates with a fixed period of 16 ms which determines our frame frequency. The negative going half of the square wave output of the astable Q_1Q_2 is passed through a differentiator and becomes the first (clock) pulse in each frame. This same output pulse of the astable multivibrator then triggers the channel one monostable timer. The pulse developed by this monostable circuit is then passed through a differentiator and becomes the second pulse of each frame. Each monostable develops its own channel pulse and then triggers the next monostable down the line. After the last monostable has triggered, a sync pause occurs until the astable multivibrator again initiates a starting pulse and the whole process repeats itself. A control or data input determines the period of each monostable and thus the time interval between each pulse in a frame.

Each of the monostable timers has identical circuitry and monostables can be added

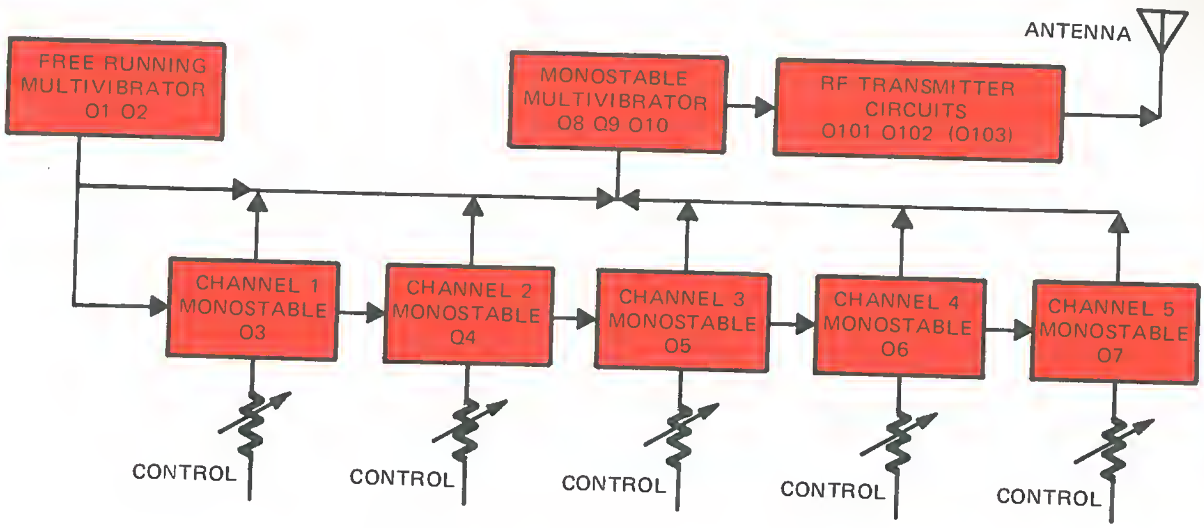


Fig. 12-6 Encoder Block Diagram

or left out to produce the number of channels to fit a specific need. Normally in this type of PPM system a minimum of about 10 channels are used. If the pulse train is upset by a noise pulse, all the channels after this pulse are erroneous until the system is resynchronized by the next sync pause. Thus, the more channels used the longer it is between sync pauses and system resynchronization.

The astable circuit is shown in figure 12-7 and has its operating periods determined by the time constants of R_2C_1 and R_3C_4.

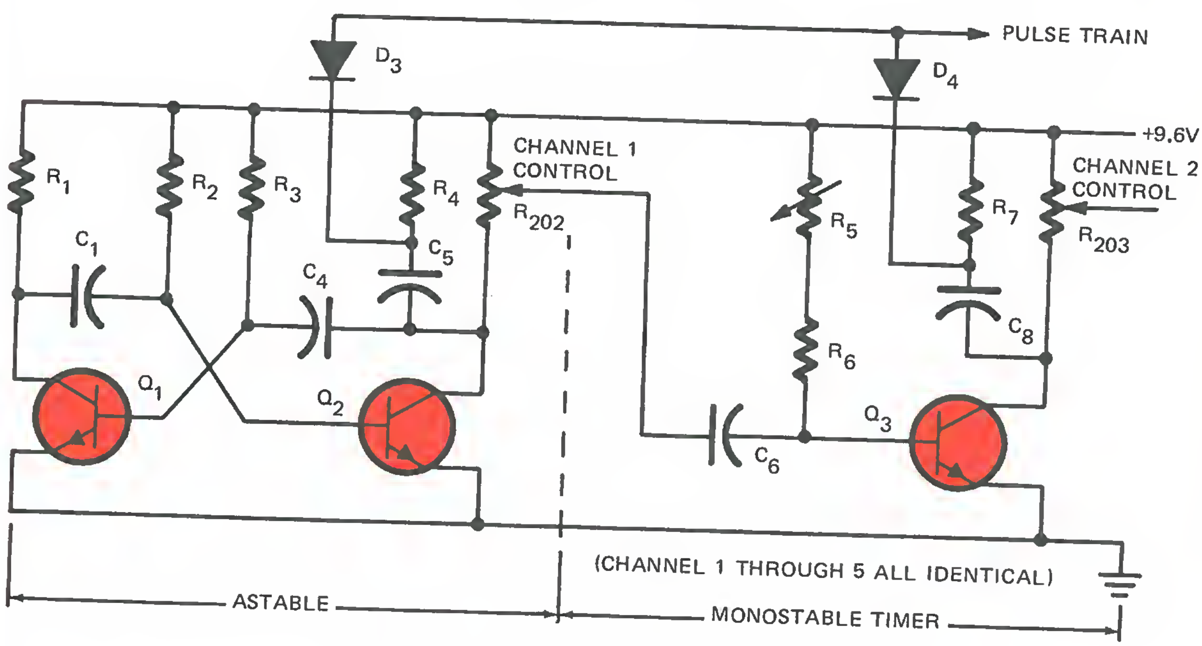


Fig. 12-7 Astable and Monostable Circuits

Since the output waveform will usually have a 50% duty cycle we normally let $R_2 = R_3$ and $C_1 = C_4$. The operating period can be determined by (for R = 111 kΩ and C = 0.1 μF)

$$T = 2(0.69)\ RC \tag{12.2}$$

$$T = 2(0.69)(1.11 \times 10^5)(1 \times 10^{-7})$$

$$T = 15.3\ \text{ms}$$

This 15.3 ms is reasonably close to the needed 16 ms considering the use of 10% components.

R_4C_5 and R_7C_8 are the differentiating networks and are connected to a common pulse train line by the decoupling diodes D_3 and D_4. These diodes are used to prevent signals from the monostable timers interfering with the operation of the astable multivibrator and vice-versa.

Q_1 and Q_2 will be switching with a 16-ms period which means each transistor will have an ON period of 8 ms determined by the time constants of R_2C_1 and R_3C_4. Q_3 will be biased ON by R_6R_5 and its base will be at approximately +0.6 volts due to the voltage drop across its forward-biased emitter junction.

The oscillation of Q_1Q_2 will develop approximately a 9-volt P-P square wave at the collector of Q_2. This pulse is connected to the base of Q_3 through capacitor C_6 and this pulse has a variable height determined by control R_{202}.

This pulse then drives the base of Q_3 negative by an amount depending upon the position of R_{202} and causes Q_3 to cut OFF. This first channel monostable timer will then appear biased as shown in figure 12-8. The base will now be biased to approximately -4.5 volts as shown by the waveform at the base in figure 12-8. As shown by this same waveform, capacitor C_6 will slowly charge through R_5 and R_6 until the base is driven to +0.6 volts and the transistor is turned ON again.

The length of the interval during which the transistor Q_3 is biased OFF is primarily

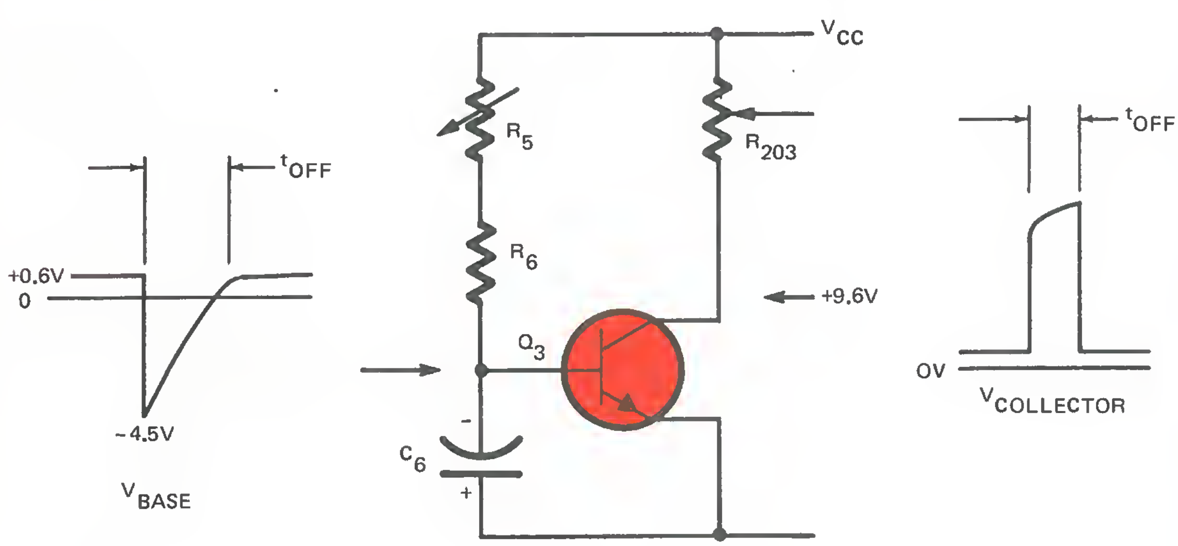


Fig. 12-8 Monostable in Cutoff Mode

determined by the amplitude of the pulse delivered by C_6. R_5 and R_6 also determine this recharge time but are fixed values and thus R_{202} is the control or data input to channel 1.

Also, when Q_3 turns ON, a negative pulse is developed at the collector of Q_3 and sent to the channel 2 monostable timer, and the same process repeats itself for each subsequent channel. Each time a transistor in a monostable is turned ON a pulse appears in the data pulse train. Remember the first pulse in the train is developed by the astable multivibrator. The resulting waveform appears in figure 12-9, and the circuit that develops each pulse is labeled above it.

Our basic pulse train is now complete, we only need to convert these pulses into pulses with a uniform width of 350 μs. We can use a monostable multivibrator as shown in figure 12-10 to accomplish this. The time constant of $R_{27}C_{32}$ will establish our desired 350 μs pulse width. A simpler monostable circuit could be used to convert the pulses, but in this system we are using Q_{10} to modulate the RF transmitter as well. So, in fact, the monostable multivibrator functions as a power pulse amplifier also.

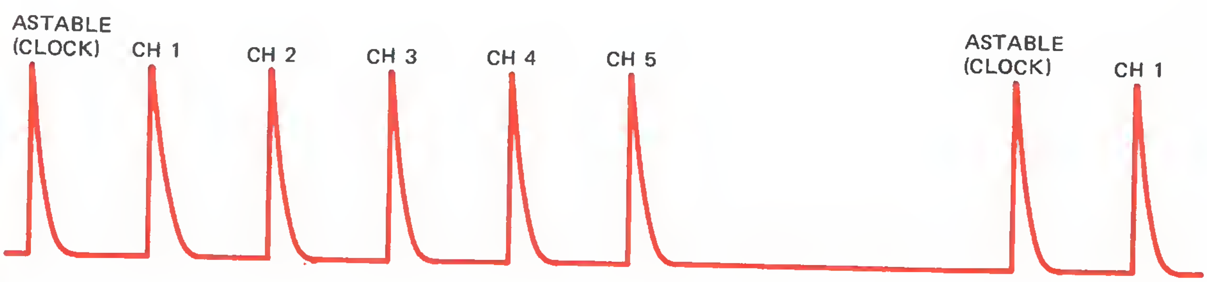


Fig. 12-9 Pulse Train

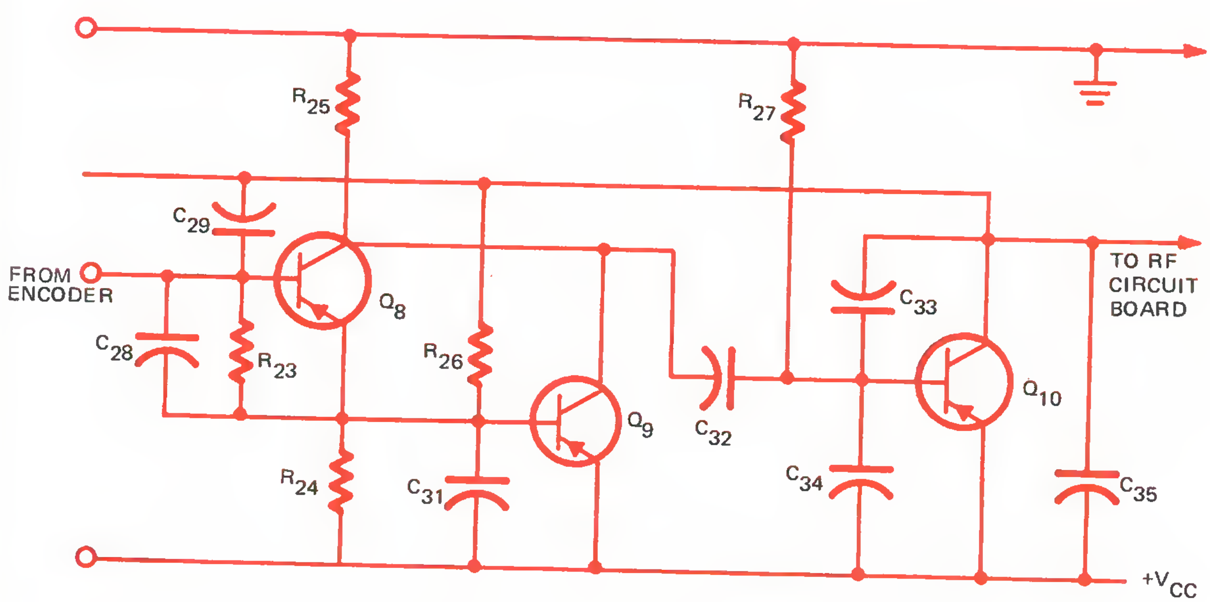


Fig. 12-10 Monostable Multivibrator - Pulse Amplifier

MATERIALS

1 Radio-control transmitter, Heath Co. GD 19-1 or equivalent
1 Oscilloscope

PROCEDURE

1. Leave the antenna fully collapsed during the following steps.
2. Familiarize yourself with figures 12-11, 12-12, and 12-13 before proceeding.

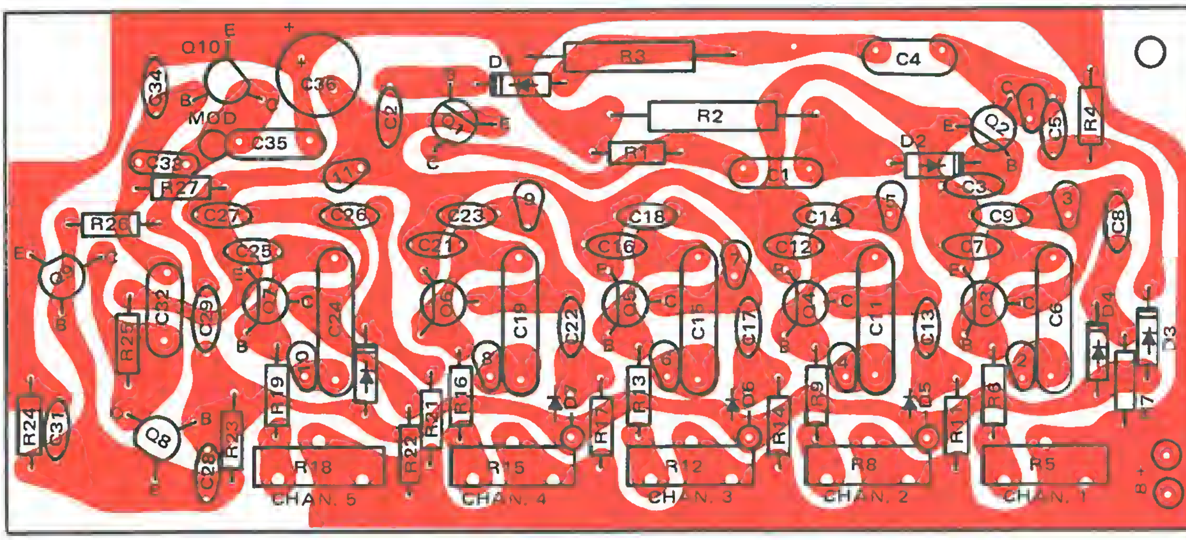


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Fig. 12-11A Encoder Circuit Board

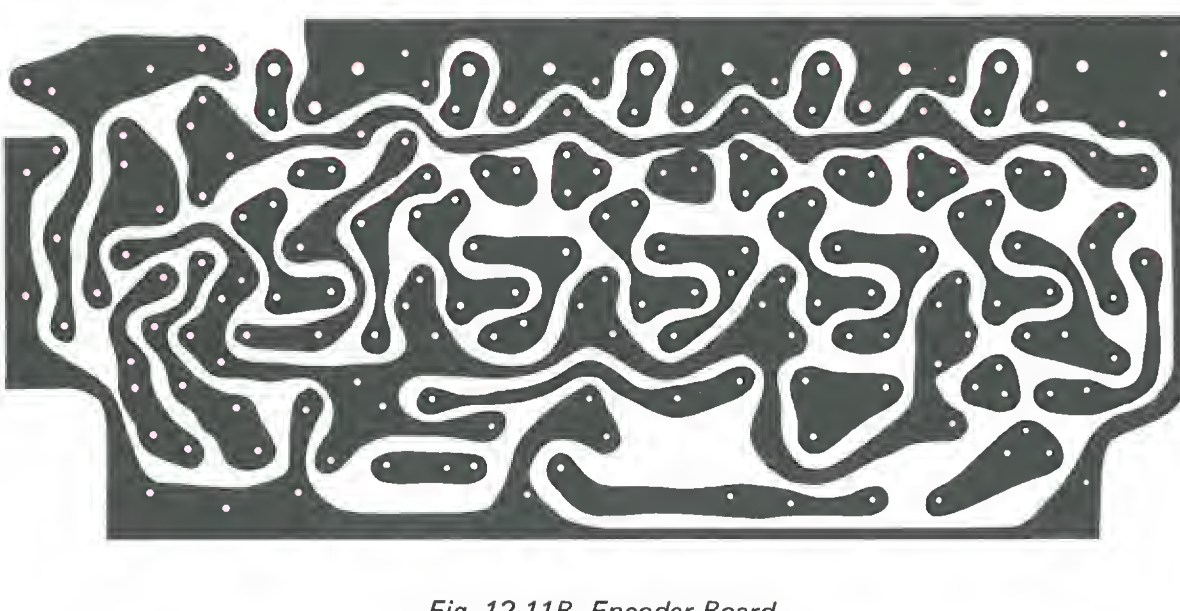


Fig. 12-11B Encoder Board

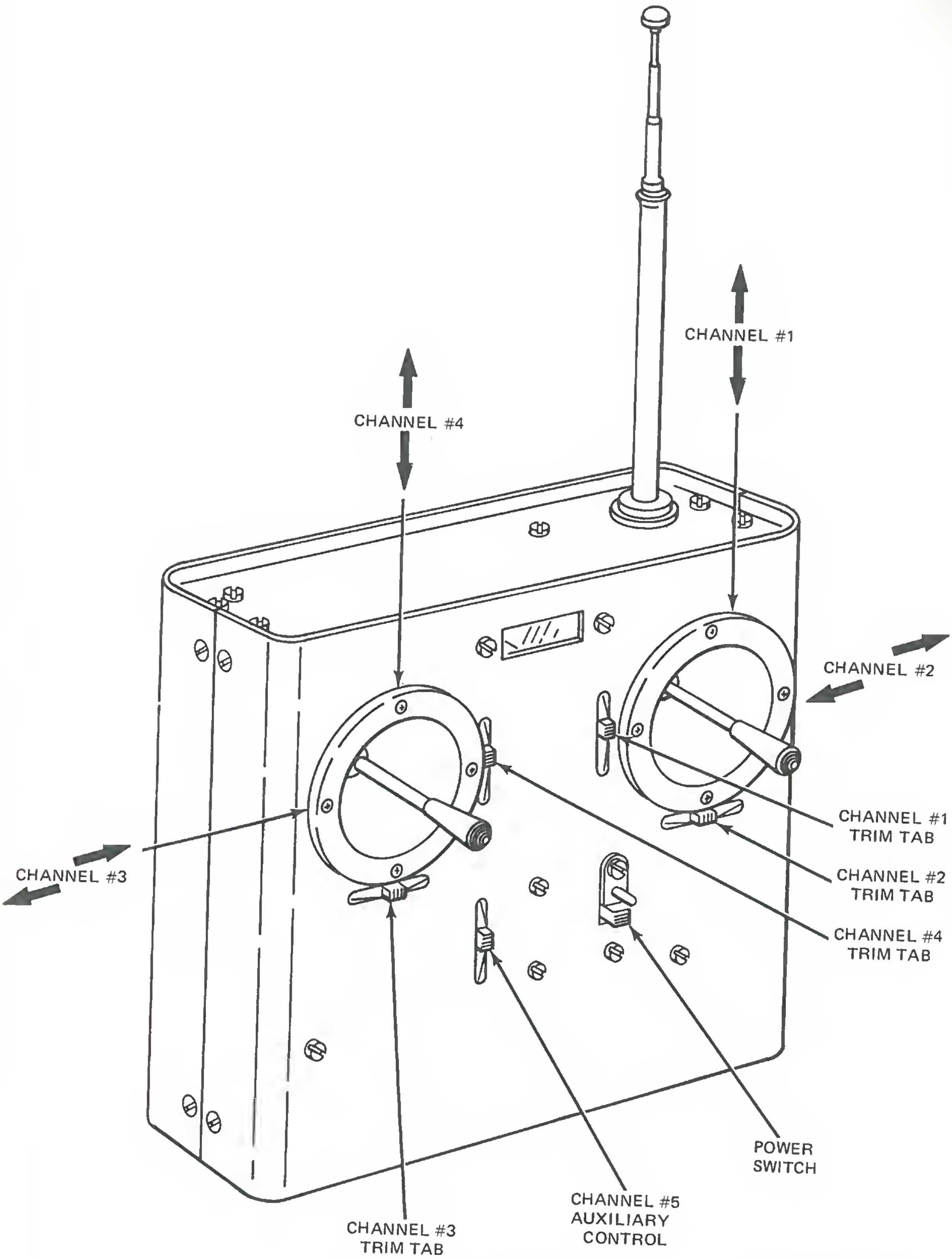


Fig. 12-12 Front Panel Layout

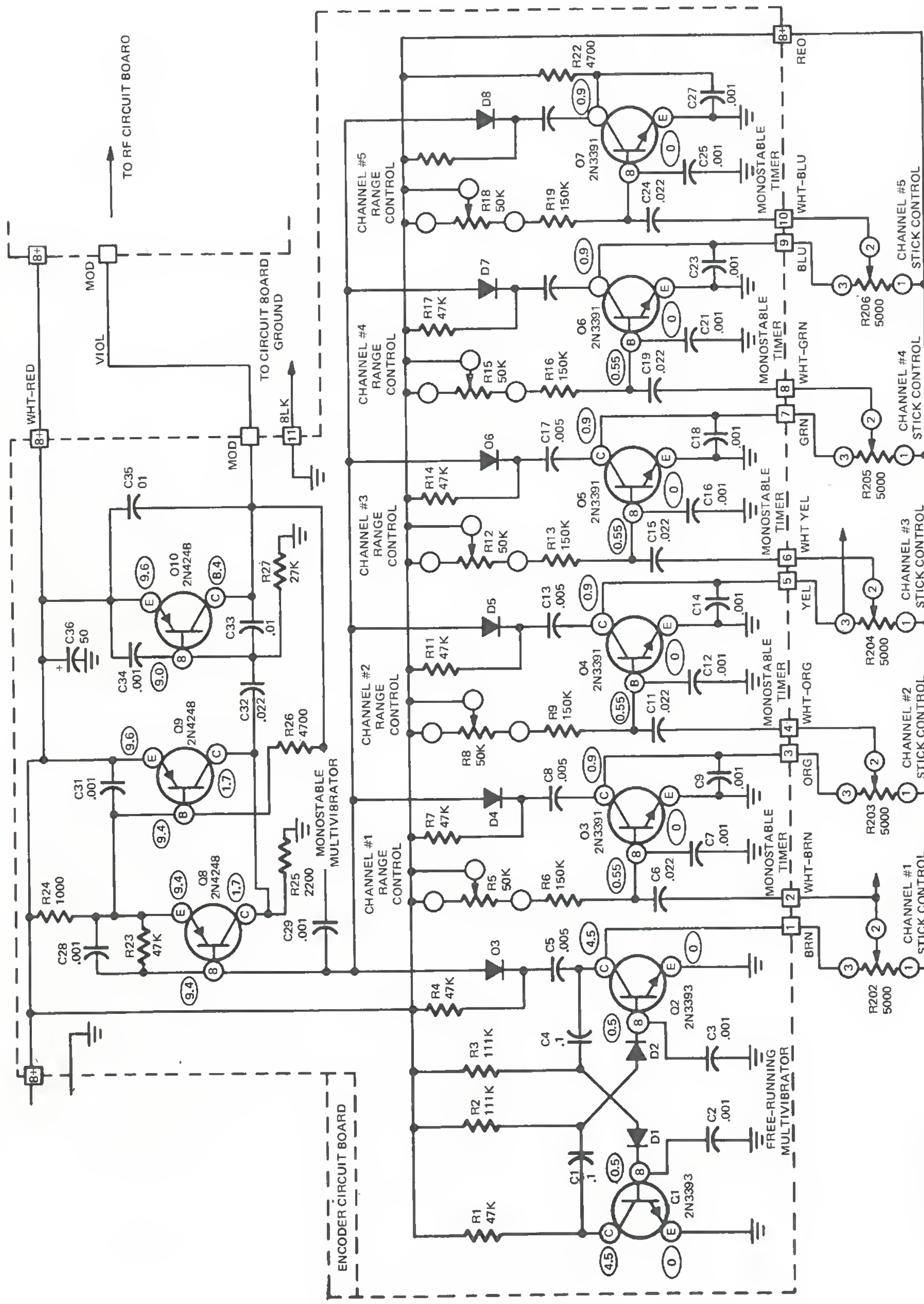


Fig. 12-13 Encoder Circuitry

(COURTESY HEATH CO.)

3. Make sure the main power switch on the transmitter front panel is OFF and then remove the rear panel by removing the screws securing it.
4. Carefully (to avoid short circuits) remove the screws securing the bottom plate. Removing this bottom plate will allow you access to the encoder circuit board as shown in figure 12-12.
5. *Case ground* will be the reference for all of the measurements to be taken during this experiment.
6. *Be particularly careful probing any printed circuit board when taking measurements.* Use a small *needle point probe* if possible. A slip resulting in the bridging of two conductors with a probe tip can cause serious damage.
7. The RF circuit board is mounted at the top of the transmitter case and the encoder board is mounted at the bottom of the case just above the battery. We will work primarily with the encoder board in this experiment.
8. Set all channel controls and trim controls to the center of their travels.
9. Since the pulse frame is referenced to the first pulse in the frame, take all measurements in relation to this first pulse. This is done by using the external sync input on the oscilloscope. Connect this external sync lead to the junction of R_{41}, C_5 and D_3. Set the triggering for a negative going signal.
10. Measure and record the following waveforms. Display two complete cycles and retain the proper time relationship between waveforms. Determine the rise and fall times and record them in the space provided. *Label all axes.*
11. Monitor the base of Q_5 and move the channel three control over its full range. Sketch the resulting waveform.

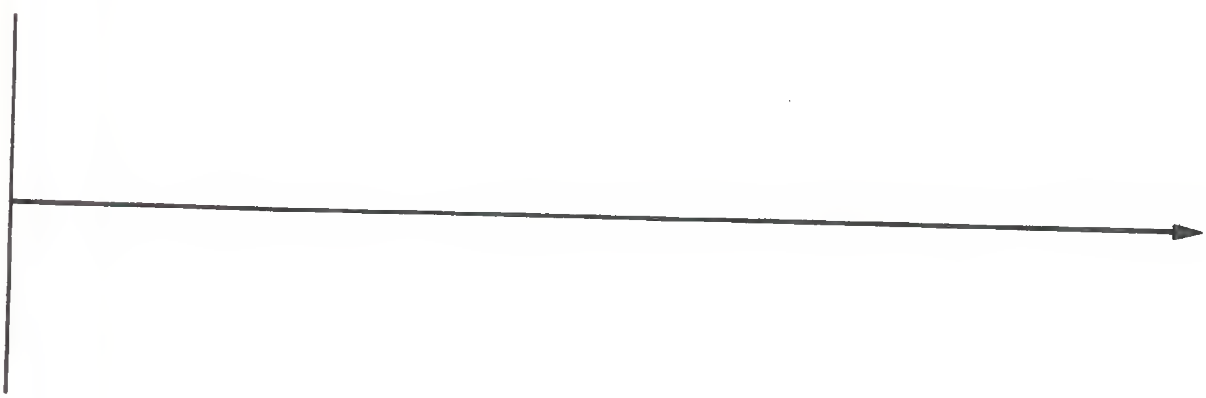


Fig. 12-14 Results from Step 11

12. Leave the channel three control at its center position. Note the position of R_{12} so that it can be returned to this position.

13. Move the channel three control through its full range and record the duration of the channel three interval. Adjust R_{12} to its maximum and minimum values and record the corresponding channel intervals.

14. Return R_{12} to its original position as noted in step 12.

15. Record the sync pause interval with all controls in their maximum channel interval position.

	Channel Interval
R_{12} Normal	
R_{12} Max	
R_{12} Min	

Sync pause interval = ________

Fig. 12-15 Results from Steps 13 and 15

ANALYSIS GUIDE. In analyzing these data you should identify each of the test points used in the experiment on the encoder circuit diagram. Then using your waveforms explain how the encoder works.

PROBLEMS

1. What causes the difference in the pulse waveform between that seen at the collectors of the monostable timer and that present at the base of Q_8?

2. What is the reason for R_6, R_9, R_{13}, R_{16}, and R_{19} in the circuits?

3. A seven-channel encoder has a frame length of 27 ms and each channel has a maximum interval of 2.35 ms. Determine the minimum interval of the sync pause.

4. Would the system described in problem 3 function correctly if its frame length were changed to 18 ms? Explain your answer.

5. What effect on the channel four pulse does a change in the position of the channel two pulse have? What effect on the channel four pulse is produced by a change in the channel five pulse?

6. Calculate the values of C_1 and C_4 if we wished to change the frame interval to 28 ms.

7. What would be the probable value of C_{15} if we wished to change channel three to have a maximum interval of 3 ms instead of 2 ms?

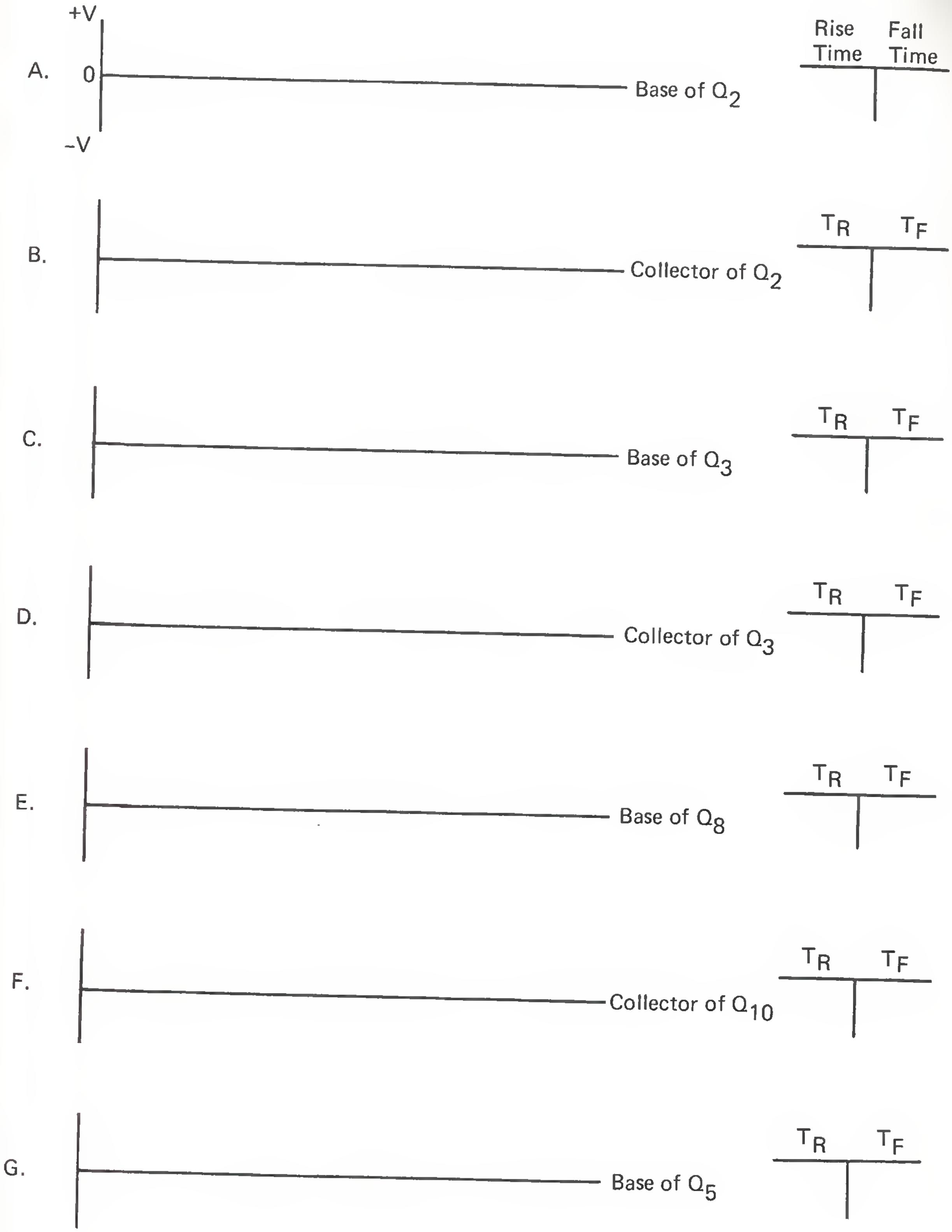


Fig. 12-16 The Data Tables

experiment 13 CERAMIC IF FILTERS

INTRODUCTION. The tuned-transformer is being replaced by the ceramic filter in many electronic circuits. In this experiment we shall investigate the principles and operation of a typical ceramic filter.

DISCUSSION. Tuned IF transformers are commonly used in superheterodyne receivers. They function as high-Q bandpass filters tuned to the IF frequency. A typical semiconductor IF amplifier circuit is shown in figure 13-1.

This circuit requires three IF transformers which must be aligned, or tuned, during the construction of the circuit. The transformers can be built as small as 0.25 inches in cubic volume. They are, however, often larger because the Q, and thus the bandwidth, of the circuit can often be improved by using a larger sized transformer.

An equivalent circuit utilizing ceramic IF filters is shown in figure 13-2. This circuit requires only two ceramic filters, F_1 and F_2, which do not require tuning and are only 0.30 inches square by 0.20 inches thick and are extremely stable and shock resistant. They have a higher Q than the usual tuned transformer and so only two ceramic filters are needed to perform the function of three tuned transformers. Due to the nature of their construction and small size it is foreseeable that they will lend themselves to inclusion in hybrid integrated circuits. This would enable a superhet receiver to be constructed consisting only of one integrated circuit, a

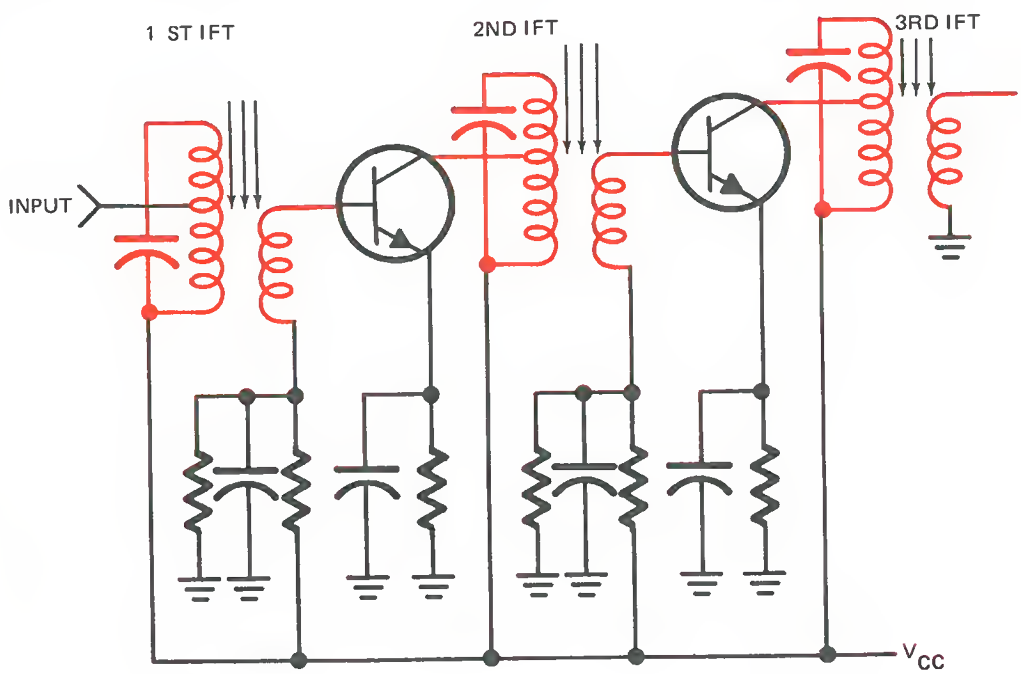


Fig. 13-1 Tuned Transformer IF Circuit

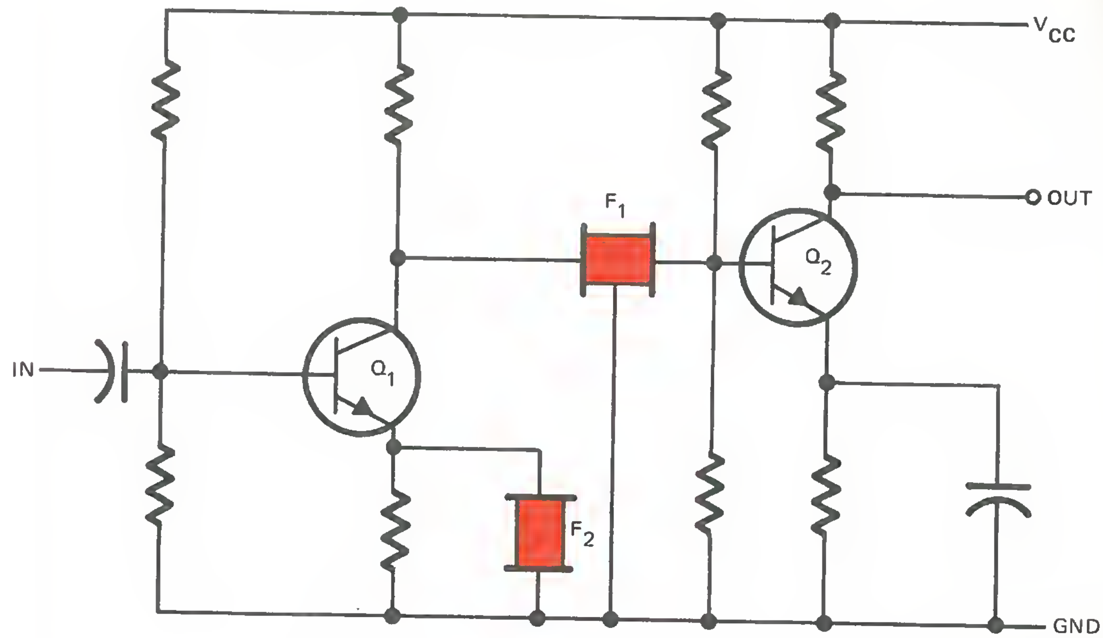


Fig. 13-2 Ceramic Filter IF Circuit

tunable RF antenna, a speaker, a volume control and a battery, possibly less than six components in all.

Ceramic filters are normally constructed from a piezoelectric element made of lead-zinconate-titanate similar to the material in a ceramic phonograph cartridge. Any piezoelectric material has three unique and related properties:

1. When a piezoelectric material is flexed it generates an electrical voltage.
2. When a piezoelectric material has electrical power applied to it, it undergoes a dimensional change.
3. A piezoelectric element will have a mechanical resonant frequency depending upon physical properties such as its dimensions and mass.

The phonograph cartridge generates a voltage as the phonograph needle flexes it. Conversely, if power were applied to the ceramic element in a phonograph cartridge, the needle would then move. Although since the typical phonograph cartridge is not designed for operation under the second set of conditions it would not be particularly efficient.

If the third property is shocked into vibration, either mechanically or electrically due to its second listed property, it will resonate or vibrate at a fixed frequency sort of like a bell or chime.

We can now utilize these three properties to better understand the operation of a ceramic filter. This type of filter has the same basic function as that of the tuned transformer and they are related as shown in figure 13-3. Both of these devices will act as a bandpass filter with an input between terminals A and C and an output taken between terminals B and C. In other words, a signal will pass from terminal A to terminal B only if its frequency is very close to the resonant frequency of the device.

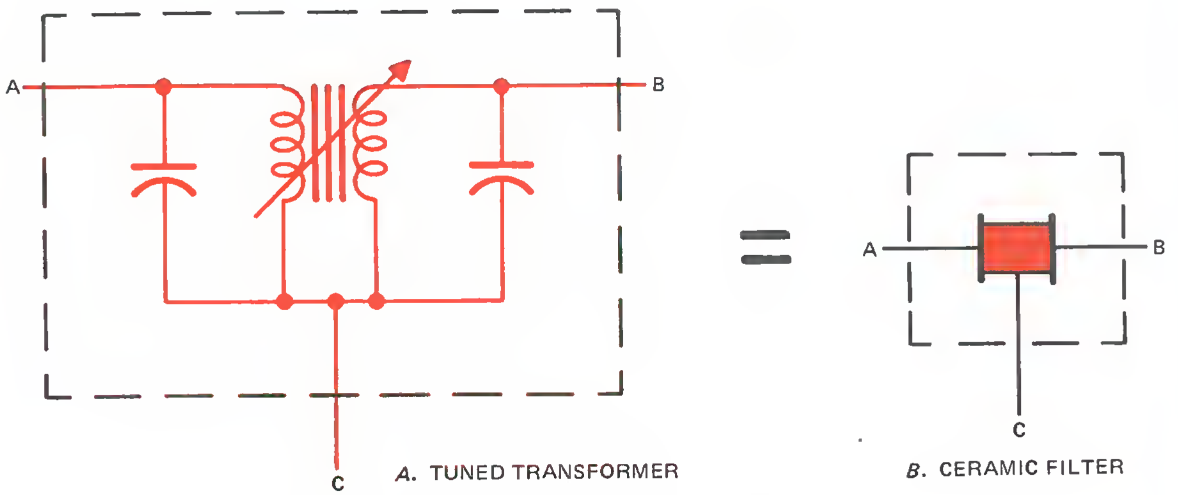


Fig. 13-3 A Tuned Transformer and a Ceramic Filter

Now let's consider the ceramic filter shown in figure 13-4.

An electrical signal applied between terminals A and C will cause the piezoelectric element to vibrate *if* the signal frequency is near the mechanical resonant frequency of the element. If the frequencies match, the element will be set into vibration in accordance to the previously stated number two property. As the element vibrates and thus flexes, it then develops a voltage between terminals B and C in accordance with property number one. As the frequency applied to the filter varies above and below the resonant frequency, less signal is passed through the filter. This action creates a curve like the one shown in figure 13-5 that is very similar to the bandpass characteristics of a tuned transformer. The Q of this filter can be fairly high and is determined by the ratio of the resonant frequency to the bandwidth

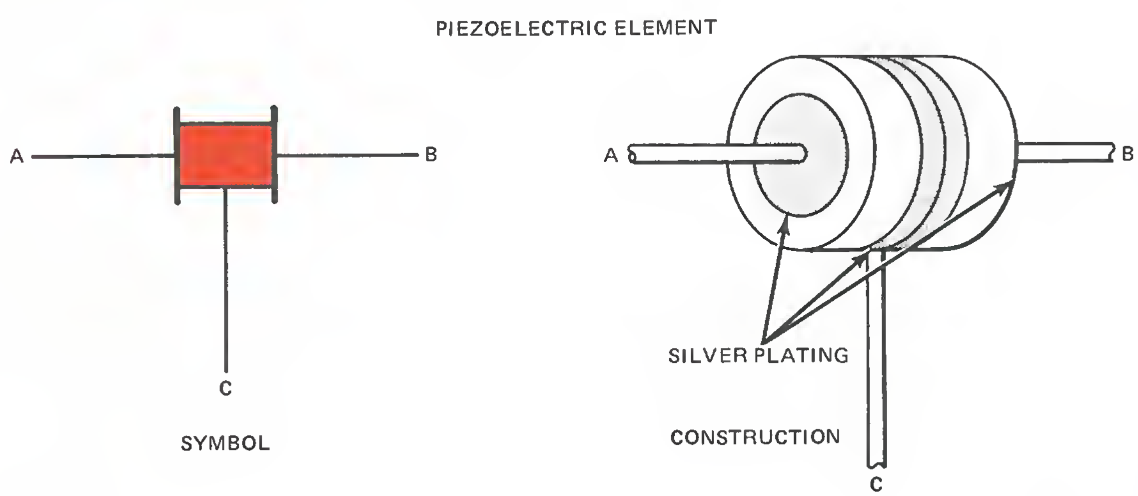


Fig. 13-4 Ceramic Filter

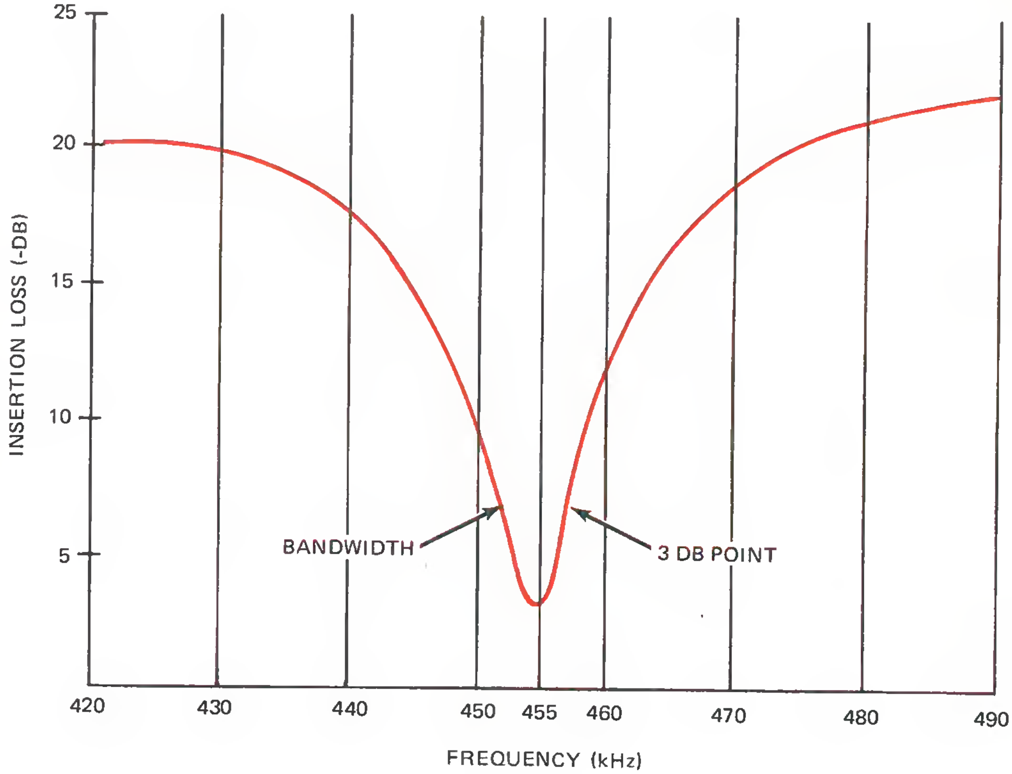


Fig. 13-5 Ceramic Filter Response Curve

at the -3db point. Typical Q's range from 50 to over 2000 depending upon the resonant frequency and application.

$$Q = \frac{\text{resonant frequency}}{\text{bandwidth}} = \frac{455 \text{ kHz}}{457\text{-}452.5}$$

$$= \frac{455}{4.5} = 101 \qquad (13.1)$$

Since the ceramic filter is a passive device and cannot amplify, it causes a signal loss when inserted in the signal path. Note that at the resonant point of 455 kHz the loss is approximately 3db or a signal voltage loss of about 30%. This loss is called an *insertion loss* and is made up for in the next amplifying stage of the circuit. The ceramic filter has specified input and output impedances and performs best when its impedances are correctly matched to that of the amplifiers associated with it. Typical impedances range from about 500Ω to about 5000Ω.

The three-terminal device just described is called a coupling filter. A related two-terminal ceramic filter called a bypass filter is also widely used and is shown in figure 13-6.

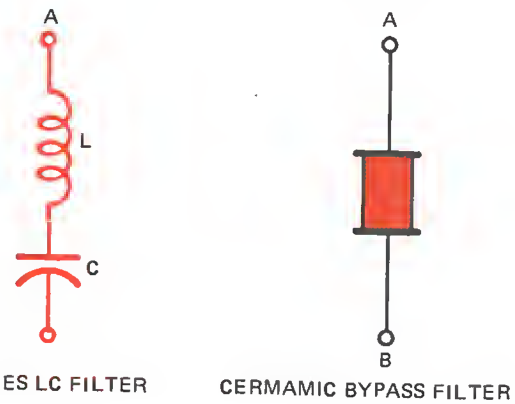


Fig. 13-6 LC and Ceramic Bypass Filters

In operation the ceramic bypass filter is that of a solid state series LC filter which has a minimum impedance at resonance and a higher impedance off resonance. For a typical application see figure 13-7.

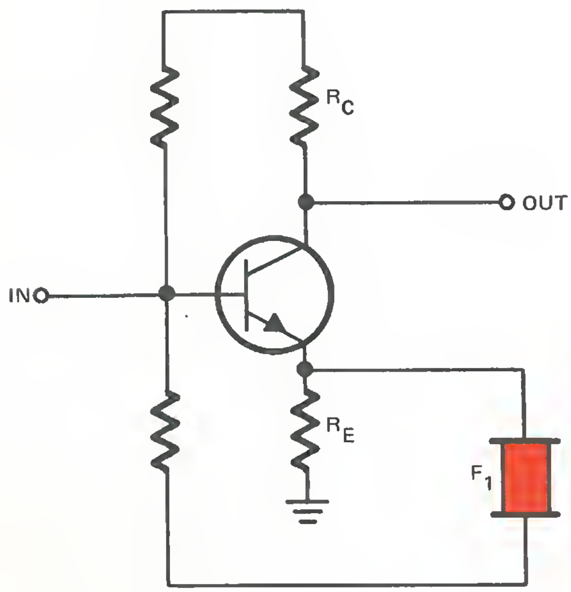


Fig. 13-7 Transistor Common Emitter Amplifier

In the above amplifier the gain is approximately described by

$$\boxed{A_V \approx \frac{Z_C}{Z_E}} \qquad (13.2)$$

From the above formula we can see that the gain of the amplifier is inversely proportional to the impedance of R_E and will reach a maximum when Z_E approaches zero. Since the ceramic bypass filter behaves much like a series LC filter having minimum impedance at its resonant frequency, the stage gain of this amplifier will be maximum at this frequency. As the Q of the ceramic filter is relatively high the circuit of figure 13-7 will have the approximate selectivity and bandwidth of an amplifier with a single tuned coupling transformer.

The ceramic coupling and bypass filters are often used in conjunction with each other in a circuit similar to figure 13-2 resulting in an economical, compact tuned amplifier for applications such as an IF amplifier.

Figure 13-8 is the schematic of a remote control digital receiver that uses three direct-coupled ceramic filters CF_1, CF_2, and CF_3. The use of direct-coupled ceramic filters results in a very high overall Q while the insertion loss is easily nullified by the high gain of the IF amplifier Q_3 and Q_4. In this circuit the IF frequency was selected as 453 MHz as can be determined by comparing the transmitter and receiver crystal frequencies.

MATERIALS

- 1 Ceramic bypass filter Clevite TF-01A or equivalent
- 1 NPN transistor, type 2N2926 or equivalent
- 2 3.3kΩ, 1/2w resistors
- 1 10kΩ, 1/2w resistor
- 1 1.2kΩ, 1/2w resistor
- 2 0.01μF, 50V capacitors
- 1 Signal generator capable of 500 kHz
- 1 Oscilloscope
- 1 DC power supply (0-40V)

CRYSTAL FREQUENCY MHZ

27 MHZ BAND		53 MHZ BAND		72 MHZ BAND	
TRANSMITTER	RECEIVER	TRANSMITTER	RECEIVER	TRANSMITTER	RECEIVER
26.995	26.542	53.100	26.3235	36.040	36.2665
27.045	26.592	53.200	26.3735	36.120	36.3465
27.095	26.642	53.300	26.4235	36.200	36.7065
27.145	26.692	53.400	26.4735	36.480	36.7065
27.195	26.742	53.500	26.5235	37.820	37.5935

	C1	C4
27MHZ	47PF	47PF
53-72MHZ	27PF	27PF

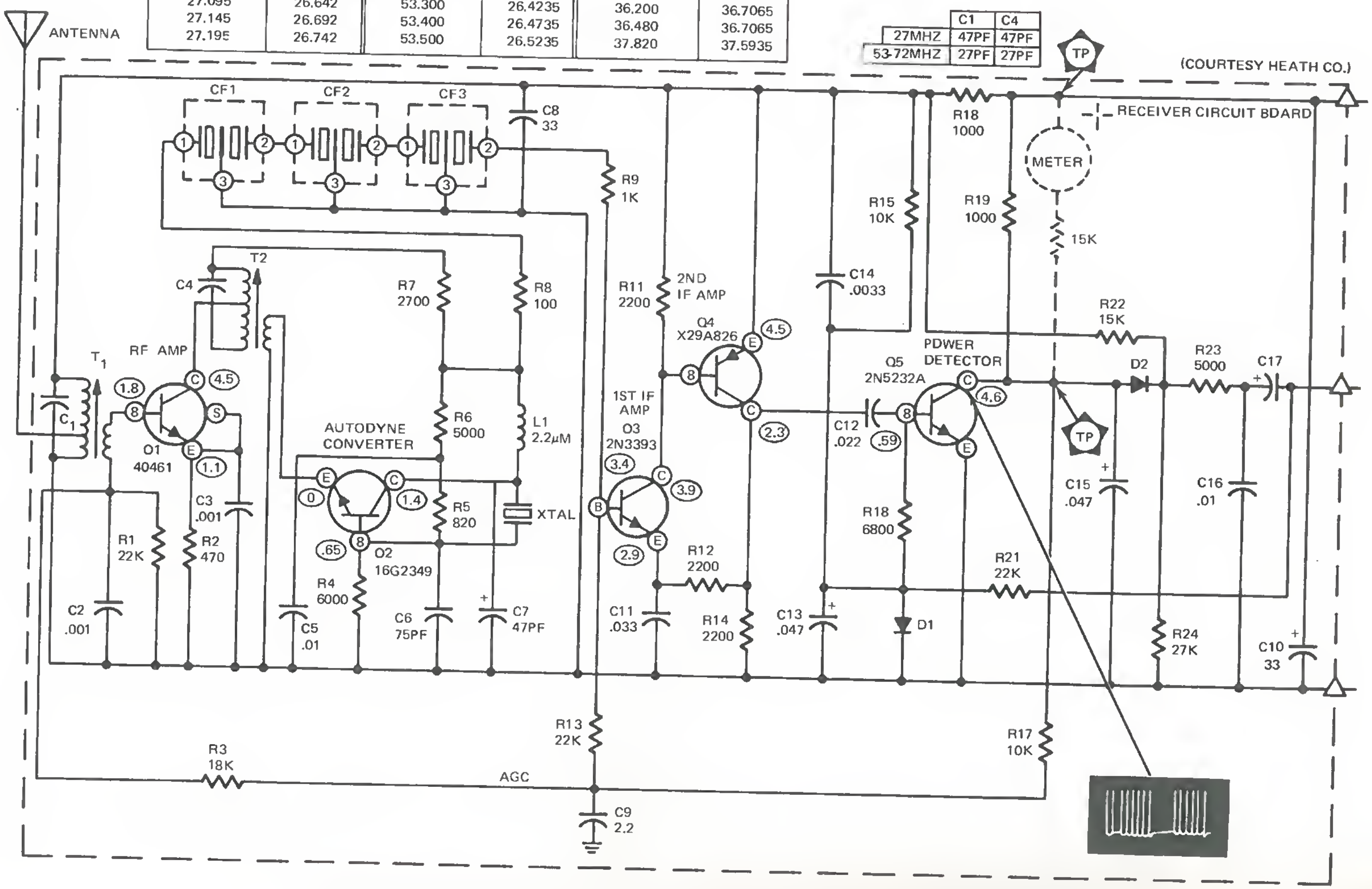


Fig. 13-8 Schematic of a Remote Control Digital Receiver

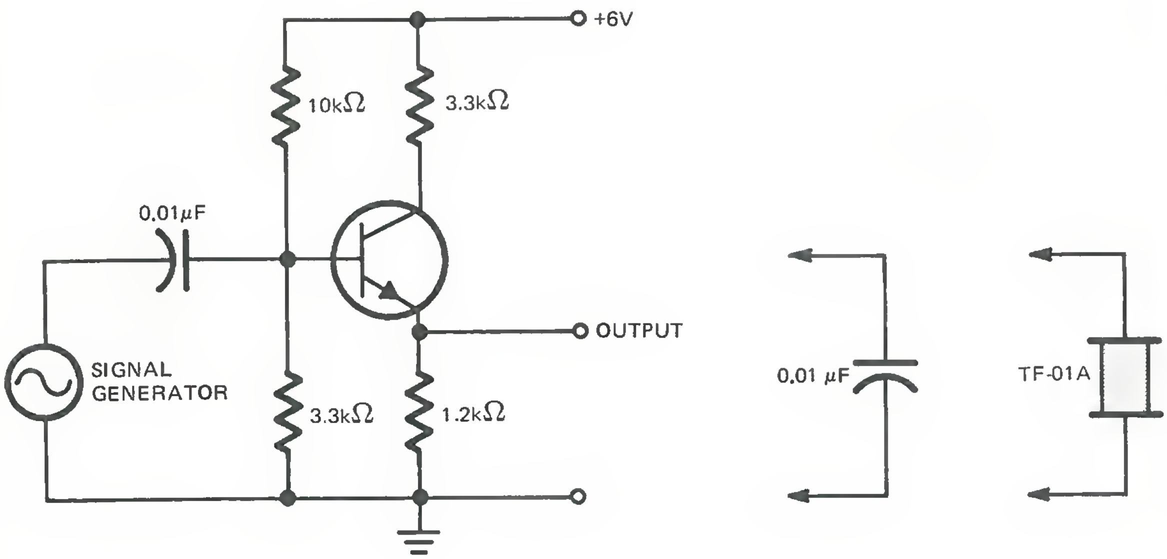


Fig. 13-9 Experimental Setup

PROCEDURE

1. Construct the amplifier shown in figure 13-9.
2. Compute the maximum current that the power supply will have to deliver to the circuit under worst case conditions. Show calculation and record.

$$I_{max} = \underline{\hspace{3cm}}$$

3. If power supply has a current limiter, set it to twice the value computed in step 2.
4. Connect the 0.01μF capacitor in parallel with the 1.2kΩ emitter resistor and turn ON the power supply.
5. Set the signal generator to 455 kHz with an amplitude of about 20 mV.
6. Monitor the output with an oscilloscope and adjust the input amplitude until the output just starts to clip the waveform. Reduce the input until the output has an amplitude equal to about 50% of the amplitude at which clipping just occurred. This will insure that the signal is not distorted more than necessary by the amplifier. Measure and record the input voltage and keep the E_{in} constant throughout the rest of the experiment.
7. By use of the variable vertical attenuator on your oscilloscope set the sensitivity so that the peak-to-peak value of the output waveform is several divisions high at 455 kHz. This will now be our reference. Change the amplitude of the signal generator as necessary to keep E_{in} constant. Do not change the sensitivity of the *oscilloscope* during the following steps.

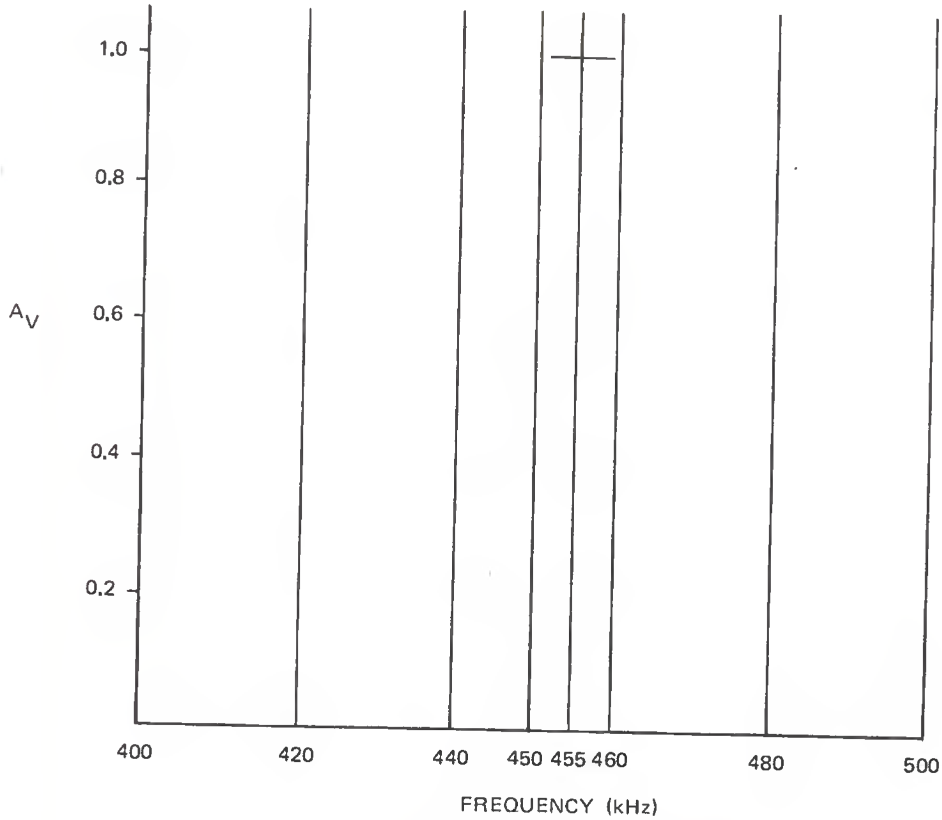


Fig. 13-10 Response Curve

8. Vary the frequency of the generator from 400 kHz to 500 kHz and record the relative gain in figure 13-10. The resulting response curve found in this step must intersect A_V = 1 at 455 kHz as it is our reference.

9. Remove the 0.01μF capacitor used in steps 4-8 and plot the response curve again on figure 13-10 from 400 kHz to 500 kHz.

10. Connect the ceramic bypass filter across the 1.2kΩ emitter resistor and plot the response curve again. Plot the curve at 2.5 kHz intervals around the resonant point for accuracy. Calibrate the scope and record E_o (max).

11. The -3db point corresponds to the gain on the curve at 0.707 of the maximum gain at resonance. Using the results from your curve, compute the Q of the circuit.

12. From the data obtained determine the maximum voltage gain.

A_V = ________

ANALYSIS GUIDE. Consider the effect on the voltage gain and on the frequency response of the three configurations you ran. Compare the three against one another. Which one do you think was the best to use as an IF amplifier? Why?

PROBLEMS

1. Changing the collector load resistor to 4.7kΩ would have what effect on A_V? How much effect?
2. In figure 13-2, what effect on the A_V would an open F_2 have? An open F_1?
3. Increasing the physical size of the piezoelectric element in a ceramic filter would have what effect on its resonant frequency? Explain.
4. Give an actual application of property number 2 of piezoelectric element.
5. What is the bandwidth of a ceramic filter with a resonant frequency of 640 kHz and a Q of 170?

experiment 14 DECODER CIRCUITRY

INTRODUCTION. A decoder is a very important part of any digital data transmission system. In this experiment we will examine the operation of a simple type of decoder circuit.

DISCUSSION. The distance between a reference pulse and a data pulse determines the value of our data in a coded pair PPM system. The reference pulse could be a clock pulse as is the case in figure 14-1, or we could refer to a previous data pulse as shown in figure 14-2. This pulse train can be sent over a wired system such as a telephone line or a coaxial transmission line. It could also be encoded on a laser beam or used to modulate a radio transmitter. The method of transmission is not important for this experiment and we can consider any of the above as being possible.

In the system we are about to examine, we will use the self-referencing system of figure 14-2. The pulse train will contain five channels of data and will have a frame length of 16 ms. One frame will appear as shown in figure 14-3.

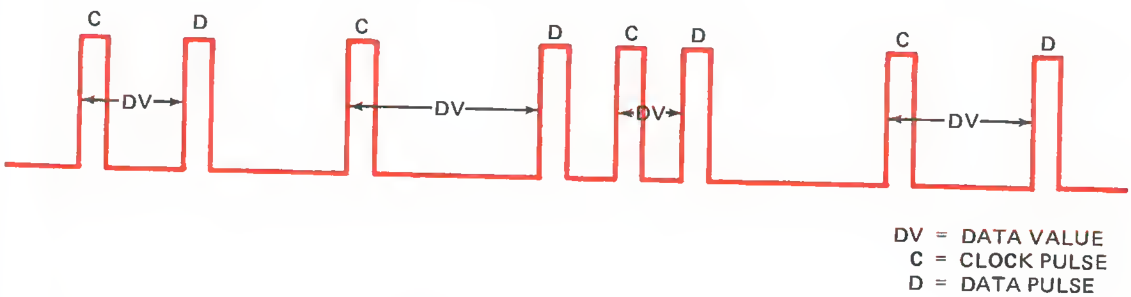


Fig. 14-1 Clock-Referenced PPM

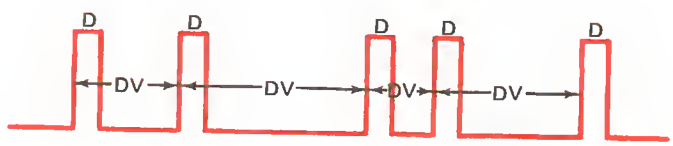


Fig. 14-2 Self-Referencing PPM

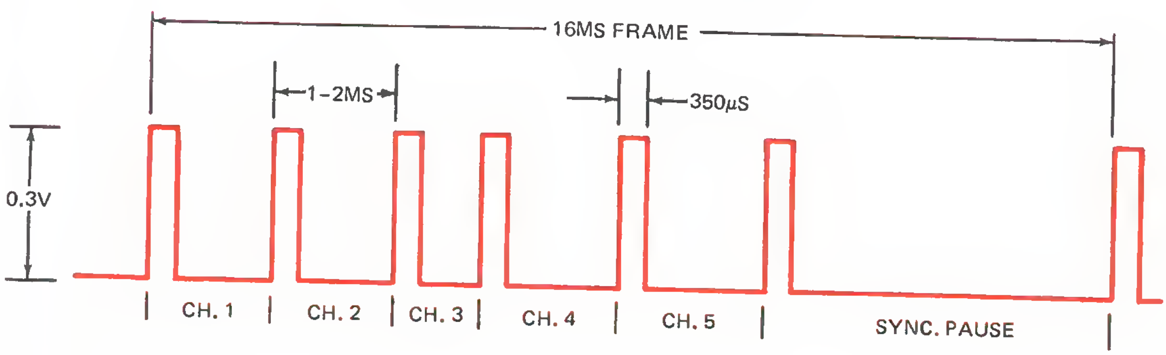


Fig. 14-3 A Five-Channel Frame

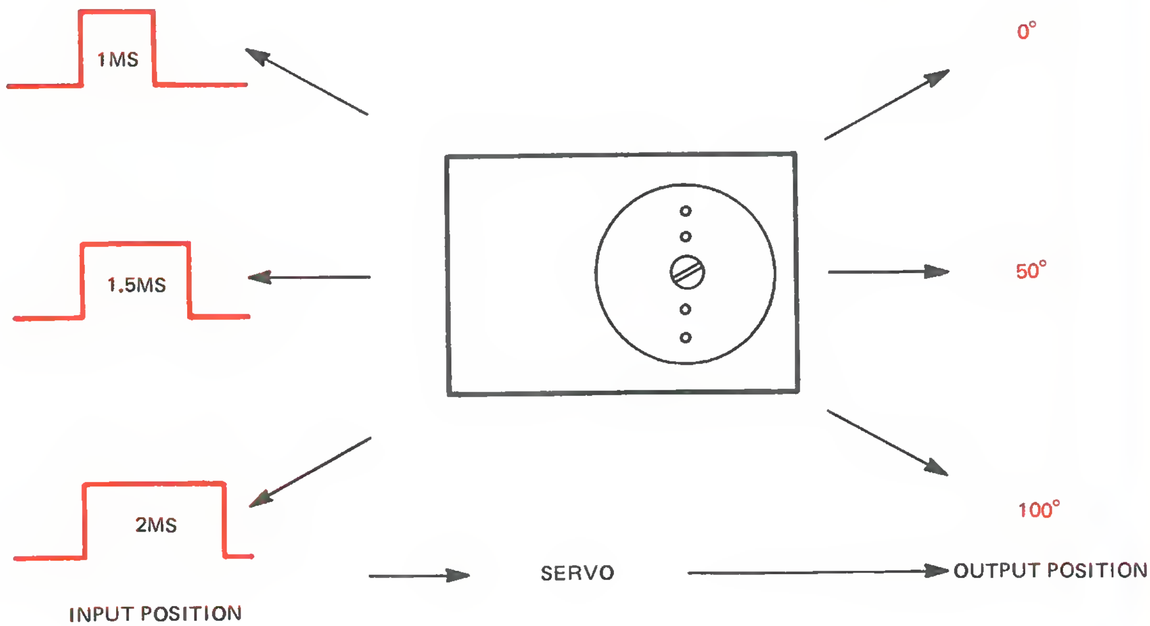


Fig. 14-4 Servo Position-Pulse Width Relationships

This PPM signal will be converted into five separate variable width pulses by our decoder circuit. The individual pulses will then be sent to each of five separate servo systems. Each servo will then convert the width of the individual pulse into a corresponding angular shaft position. For instance, a 1-ms pulse will represent an angular position of 0°. A 1.5-ms pulse produces an angle of 50° and a 2-ms pulse produces an angle of 100°. This relationship is shown pictorially in figure 14-4.

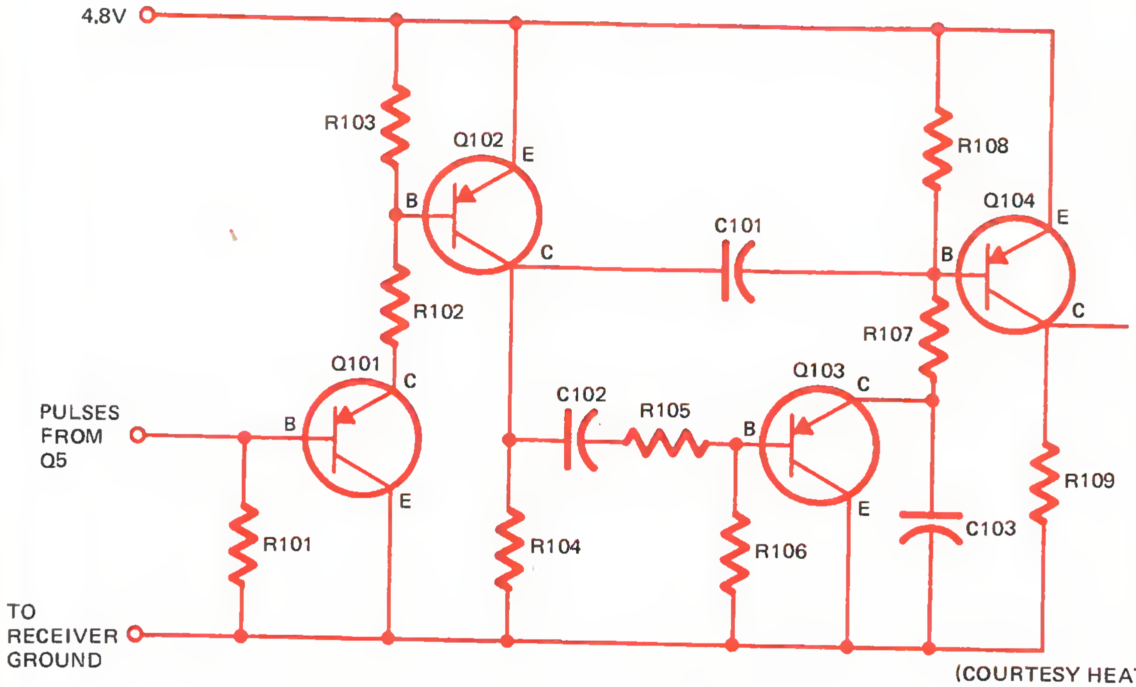
The sync pause will be used to reset the decoder and keep it in step with the encoder pulse train.

The decoder we will be using requires an input signal of about +0.3V P-P. This signal is amplified to about +4.8V P-P by the complementary pulse amplifier Q101 and Q102 shown in figure 14-5.

A multi-stage complementary amplifier is commonly used in pulse applications because it draws power only when it is actually amplifying a pulse and does not draw power when it is idle.

Without an incoming signal R101 biases Q101 OFF which in turn allows R103 to bias Q102 OFF thus placing the output at near ground potential. When a positive pulse is applied to the input, Q101 turns ON which turns Q102 through R102 and the output raises to about +4.8V. Thus current is drawn only during the presence of a positive pulse at the base of Q101.

The signal is then coupled to Q103 which serves to detect the sync pause. The signal from Q102 is also coupled to Q104 which is normally held OFF by R108 and cannot pass this train of positive going pulses as they only serve to turn it further OFF. However, the first pulse of a frame turns ON Q103 which then clamps R107 to ground, discharging C103 and turning ON Q104. Q104 being now biased ON can pass the pulse train



(COURTESY HEATH CO.)

Fig. 14-5 Complementary Pulse Amplifier and Sync Circuit

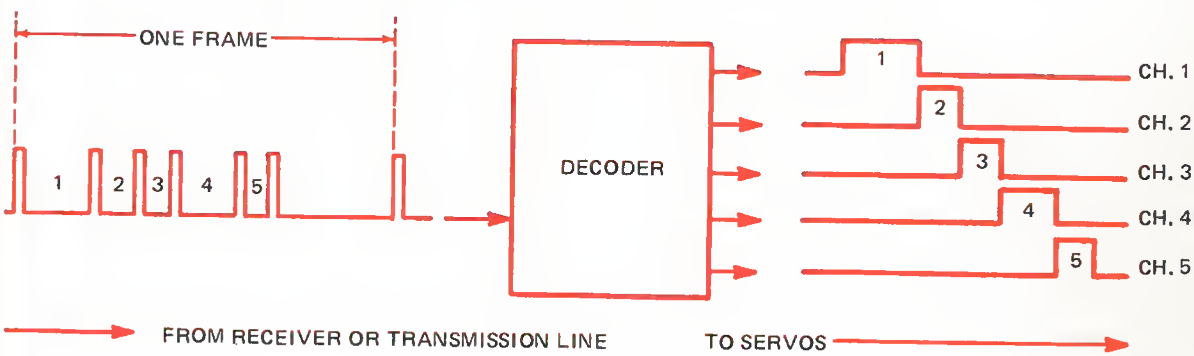


Fig. 14-6 Decoder Waveforms

as each pulse now momentarily turns it OFF. When the sync pause occurs Q103 remains OFF for 6 ms to 11 ms and C103 charges through R107 and R108 and then turns OFF Q104.

The time constant of R107, R108 and C103 is approximately

$$(R107 + R108)\,(C103) = \tau$$

$$(20\ k\Omega)\,(0.22\ \mu F) = 4.4\ ms \qquad (14.1)$$

This period is much longer than the maximum interval of 2 ms between data pulses and thus Q104 remains ON between them. Whenever Q104 is OFF for the duration of a sync pause, the remaining part of the decoder resets itself.

The decoder takes the pulse train and separates it into 5 pulses using separate circuits. Each pulse now having a width equal to the interval of one of the 5 data channels as indicated in figure 14-6. Notice that each

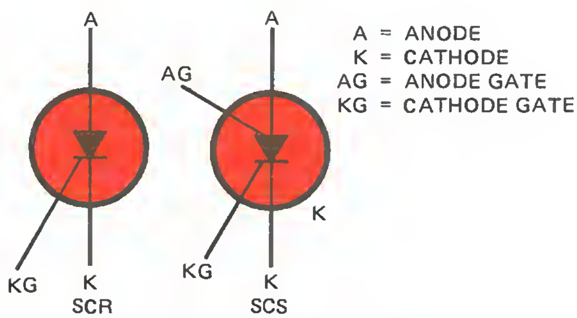


Fig. 14-7 SCR and SCS Symbols

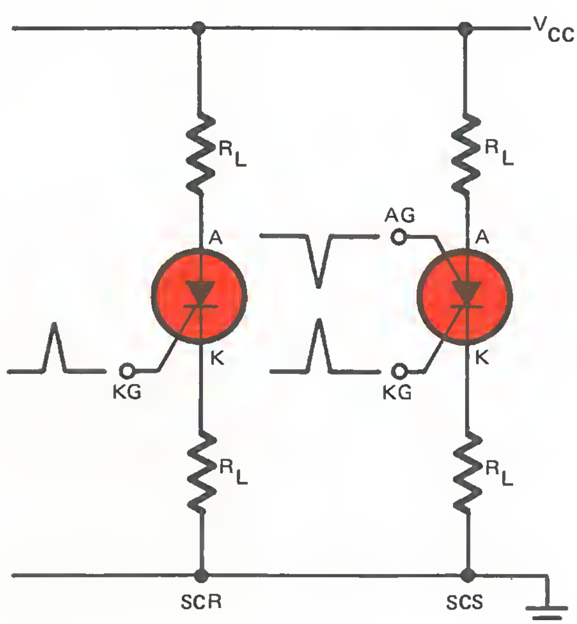


Fig. 14-8 SCR and SCS Turn-On Pulses

output pulse starts at the trailing edge of the previous pulse.

This decoder uses a device known as an SCS (Silicon Controlled Switch). An SCS is similar to an SCR but has an anode gate as well as a cathode gate as shown in figure 14-7.

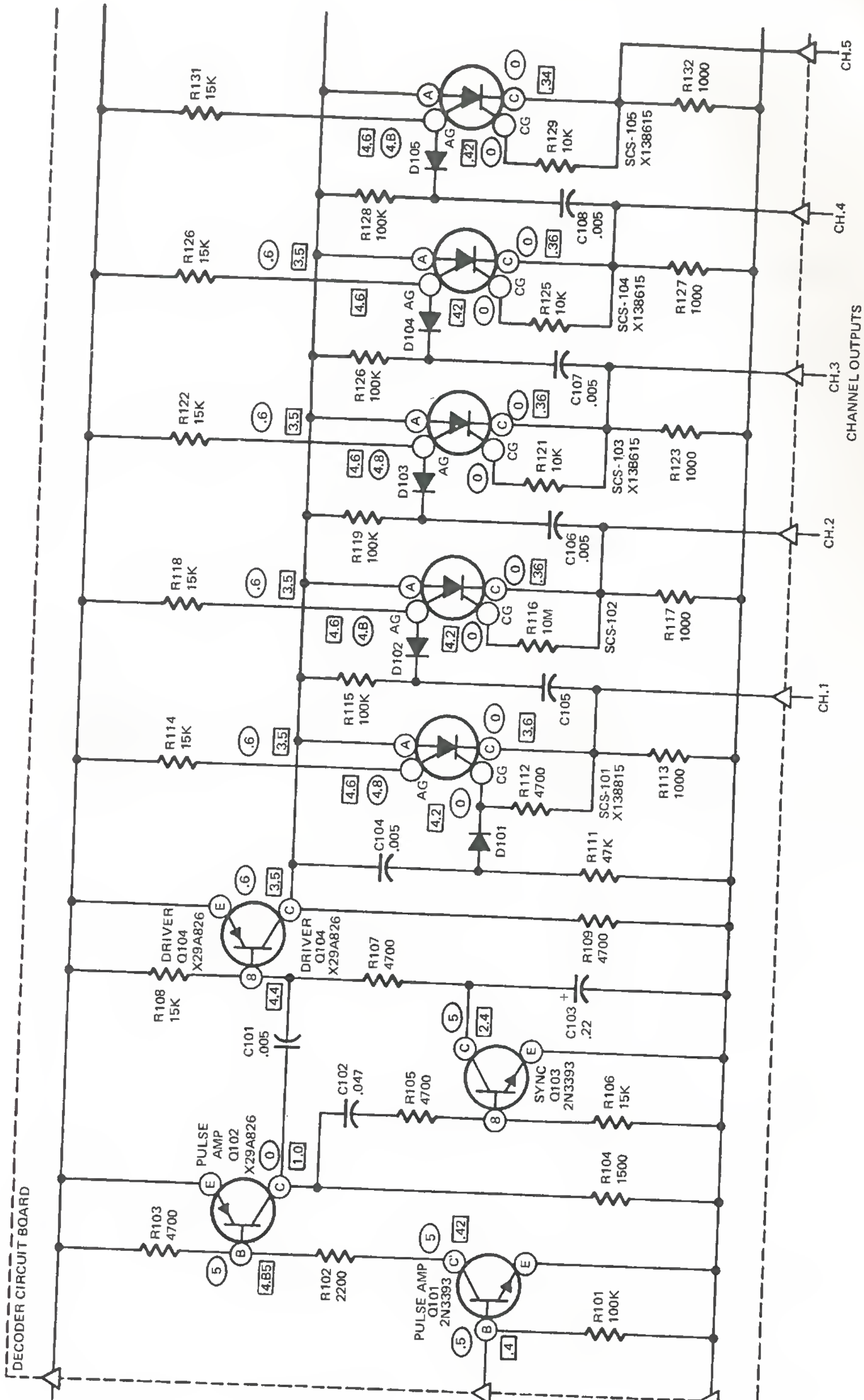
An SCR is a diode that requires a signal be applied to its cathode gate before it will conduct in the forward direction. Once in a forward conducting mode, it cannot be turned OFF until the anode to cathode voltage drops below the minimum holding voltage (V_h).

The SCS has a somewhat similar operation and can be turned ON and OFF by application of suitable signals to its anode gate and cathode gate. Like the SCR, reducing the anode to cathode voltage of an SCS momentarily below V_h will also cause it to turn OFF. The signals applied to an SCR or SCS need only be short-duration pulses, because once the device changes state it will remain in that state even if the trigger pulse is removed. Examples of basic SCS and SCR circuits are shown in figure 14-8 along with the pulse signals required to turn them ON. Note that both devices can be turned ON by positive-going pulses applied to their cathode gates and that the SCS can also be turned ON by a negative-going pulse applied to its anode gate. The load resistors can be in either the anode or cathode circuit or both. Once the SCR is turned ON it remains ON until the anode-to-cathode voltage is dropped below V_h. The SCS will turn off when the anode-to-cathode voltage is dropped below V_h, however, the SCS can also be turned OFF by applying a negative pulse to its cathode gate or a positive pulse to its anode gate. The pulses required to turn OFF the SCS require more power than the turn ON pulses but in contrast to the SCR *the SCS can be turned OFF with its anode and cathode gates.*

The SCS is much more versatile than the SCR but is not presently available in higher power units like the SCR.

We will use an SCS circuit to convert our channel pulses in a frame into individual pulses to drive our servos.

The circuit will turn on one SCS at a time for the length of a channel interval and is commonly called a ring counter. The complete circuit is shown in figure 14-9.



(COURTESY HEATH COMPANY)

Fig. 14.9 The Decoder Circuit

Note that the circuits of SCS-101 through SCS-105 are basically the same, so the circuits of just SCS-101 and SCS-102 can be used to describe the total operation. The simplified circuit is shown in figure 14-10.

According to equation 14.1, Q104 will remain OFF anytime the pulse train ceases for more than about 4 ms. Since Q104 supplies the anode voltage to the SCS's, they will be in the OFF condition also. With the SCS's OFF all the servo inputs will be at near ground potential.

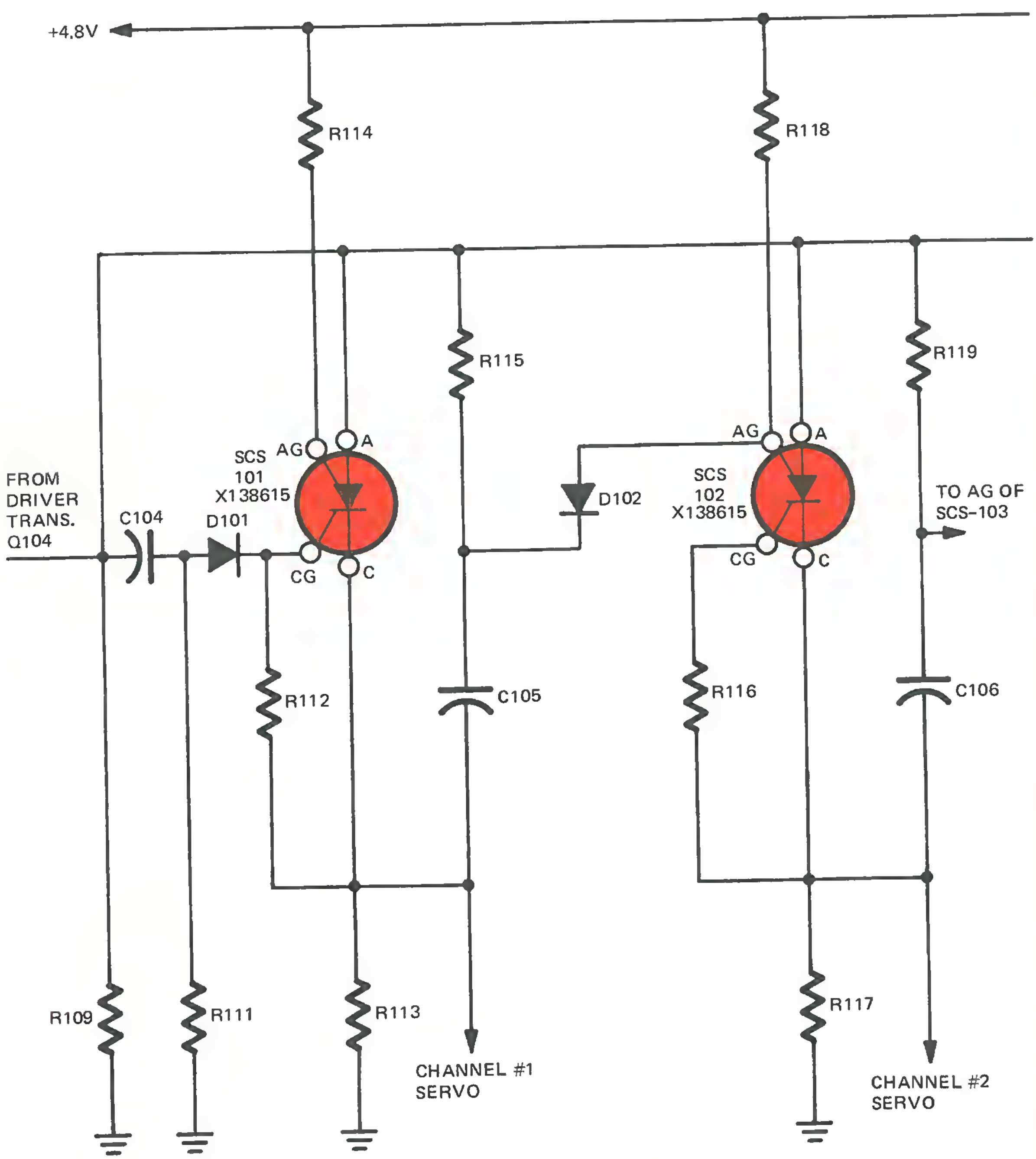


Fig. 14-10 The SCS Circuitry

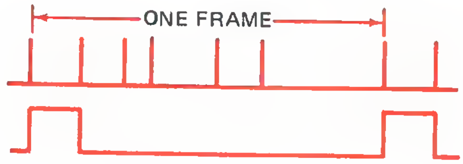


Fig. 14-11 Decoder Input-Output Relationship for Channel 1

When a pulse train arrives Q104 turns ON which supplies anode voltage to all of the SCS's and prepares them to conduct.

The presence of a positive-going data pulse at the collector of Q104 is passed through the differentiator composed of C104 and R111, through D101 and turns ON SCS-101. When SCS-101 is in its ON state, its cathode will approach +4V and apply this signal to the channel 1 servo as shown in figure 14-11.

The SCS will then stay ON until the second pulse of the frame momentarily turns Q104 OFF. This momentarily cuts OFF the anode voltage of SCS-101 turning it OFF and its cathode returns to ground potential. Thus a pulse with an amplitude of +4V and a duration equal to the interval between the first pulse and second pulse in the frame has been supplied to the channel 1 servo.

A negative going pulse is generated as the cathode of SCS-101 returns to ground. This pulse is passed through a differentiator comprised of C105 and R115 and through D102 turning ON SCS-102 by means of a negative pulse applied to its anode gate. As SCS-102 turns ON, its cathode raises to 4V forming the leading edge of the pulse to the channel 2 servo as shown in figure 14-12. The third pulse in a frame then turns SCS-102 OFF, forming the trailing edge of the pulse to channel 2 and also turns on SCS-103.

Each successive pulse turns OFF an SCS in the ON condition and turns ON the following SCS until the sync pause arrives and turns OFF *all* the SCS's resetting the SCS line for the next frame. If an erroneous noise pulse appears during a frame, say due to noise, the output pulse of the channels following the pulse will also be erroneous. However, this condition can only last for the duration of a frame, 16 ms, before a sync pause arrives and sets things straight.

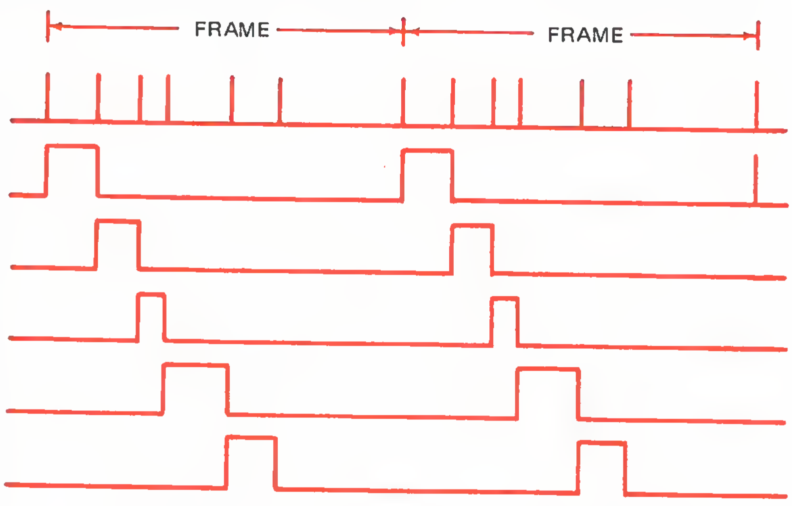


Fig. 14-12 Decoder Input-Output Relationship for All Channels

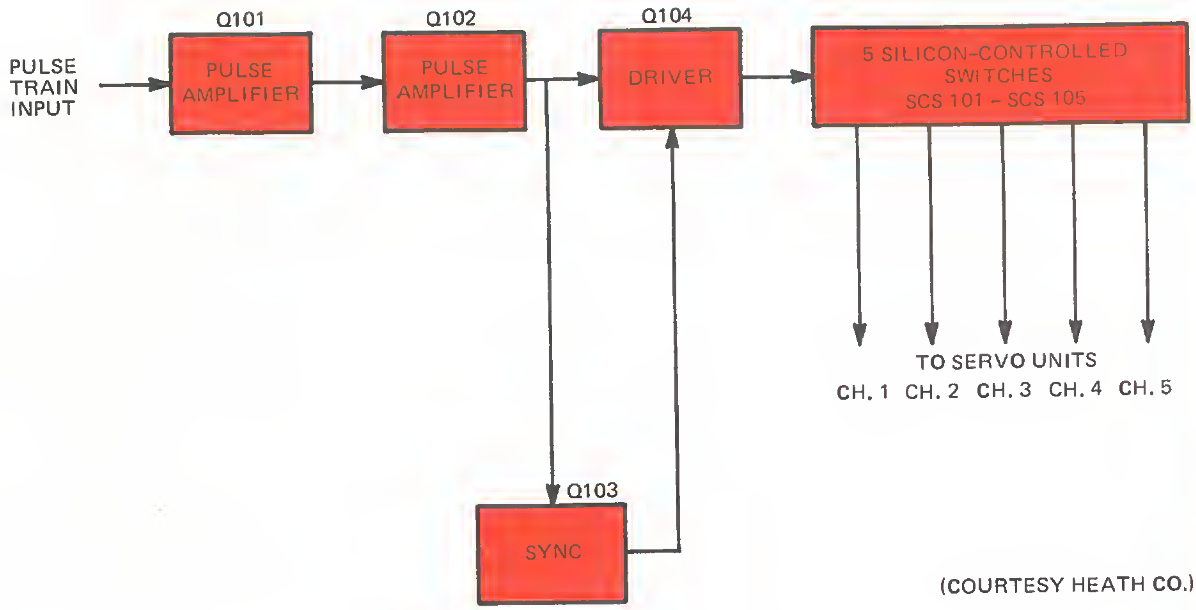


Fig. 14-13 Decoder Block Diagram

The number of channels served by a decoder can be increased or decreased simply by adding or subtracting SCS stages identical to the one employing SCS-105 in figure 14-9.

Returning to figure 14-10, the resistors R118 and R116 supply reverse bias to the anode gate and cathode gates respectively to insure that they do not turn ON as the anode voltage is applied to SCS-102. The diode D102 allows only negative going pulses to be applied to the anode gate. Each SCS stage has similar components which have the same function. A block diagram of this decoder system is shown in figure 14-13.

MATERIALS

- 1 Radio control transmitter (Heath Co. model GD 19-1 or equivalent)
- 1 Radio control receiver (Heath Co. model GD 19-2 or equivalent)
- 1 Receiver battery pack and wiring harness (Heath Co. model GD 19-3 or equivalent)
- 1 Oscilloscope

PROCEDURE

1. Refer to figure 14-14 and figure 14-9 during this procedure. The Receiver plug color code is as follows:

Channel No.	Color Code
1	Brown
2	Orange
3	Yellow
4	Green
5	Blue

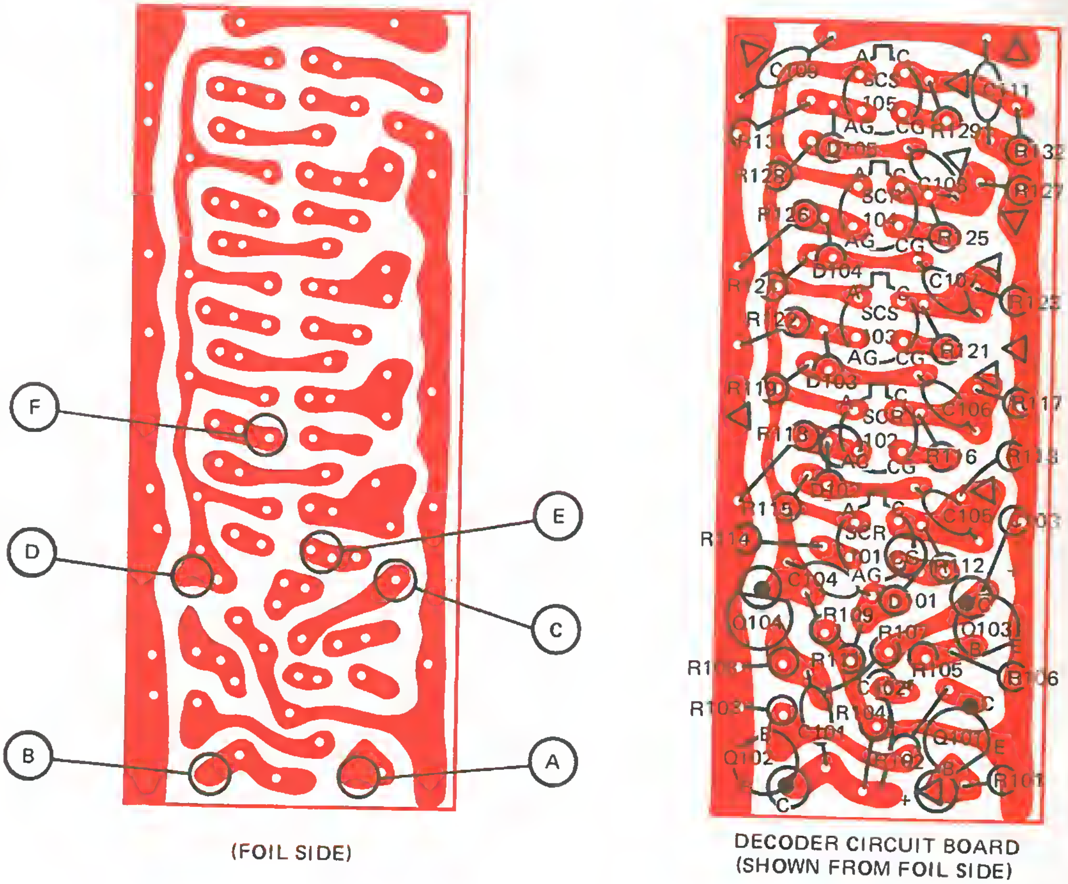


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Fig. 14-14 Decoder Circuit Board

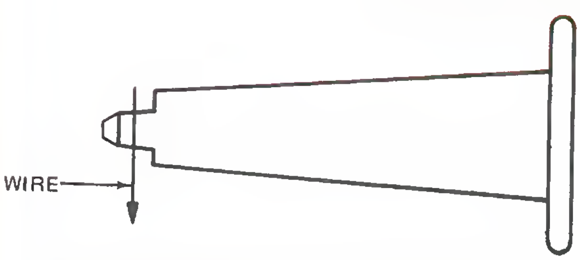


Fig. 14-15 Wire Probe Tip Extension

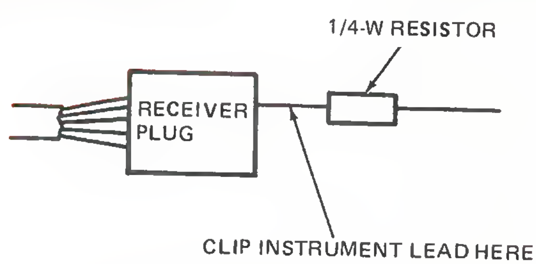


Fig. 14-16 Plug Connection

2. Make all measurements with respect to ground (black lead). *Have the channel controls set to their center positions. Place channel 4 control at its minimum interval position.*

3. Use a needle point probe when taking measurements on a printed circuit board. Use very light pressure; this board is only 0.032 in. thick. If a needle point probe is not available, attach a piece of hook-up wire to the probe tip as shown in figure 14-15.

4. Insert the lead of any 1/4-watt resistor into the receiver plug and attach your clip lead to the lead of the resistor when necessary. See figure 14-16. *Do not try to force a larger wire into the plug* or it will be deformed or break the plug.

5. Sketch the waveform, determine and record the risetime, falltime, amplitude and pulse duration *for one pulse* at each of the points in the first data table. Label the axes on the waveforms.

TEST POINT	WAVEFORM	RISETIME	FALLTIME	AMPLITUDE
Base of Q101				
Collector of Q102				
Collector of Q103				
Collector of Q104				
Cathode Gate of SCS-101				
Anode Gate of SCS-102				
Ch. #1 Output				

Fig. 14-17 The First Data Table

6. Set up the oscilloscope to sync on the leading edge of the channel 1 output pulse. Connect the sync lead to the receiver plug using the method shown in figure 14-16.

7. Sketch, in proper time relationship, the signal at the collector of Q102 and the five channel outputs. Use the second data table for these sketches. *Show one complete frame and label the axes.* Hint: Sync on channel 1 output.

Collector Q102	
Ch. 1	
Ch. 2	
Ch. 3	
Ch. 4	
Ch. 5	

Fig. 14-18 The Second Data Table

8. Move the channel 5 control to its maximum interval position. Sketch the resulting waveforms on the table in step 6 in dotted lines.

9. Calculate the time constant of R115 and C105. Compare this with the observed fall time of the waveform in step 5 at the anode gate of SCS-102.

10. Using the waveform taken at the collector of Q104 and label the portion of the waveform causéd by C103. Also, label each channel data pulse.

ANALYSIS GUIDE. In analyzing the results of this experiment, discuss the time constants concerning C103 and its purpose in the circuit. Does its time constant determined from its waveform agree with its value calculated in the discussion? Discuss also the effect of frame length and sync pause interval when channels are added or subtracted from the system.

PROBLEMS

1. Calculate the time constant of the differentiator feeding the cathode gate of SCS-101.

2. Draw the schematic containing *all* the necessary components to convert the decoder to six channels. Label all inputs and outputs.

3. Would the circuit change in problem 2 affect the length of the sync pause if no changes were made in the transmitter?

4. What is the function of R101?

5. What would be the pulse width necessary for an output position of 86° in figure 14-4?

6. What effect would a shorted SCS-102 have on system operation? What channels would be affected?

7. How could an open D105 affect the system?

experiment 15 DIGITAL SERVO

INTRODUCTION. In pulse control circuits we often use a circuit that will translate pulse durations into a positive or negative voltage to drive a servo motor. In this experiment we will examine a circuit which performs this function.

DISCUSSION. Suppose we have a series of pulses that have an interval of 6 ms and vary in duration from 1 ms to 2 ms. A servomotor is to receive these pulses and position its output shaft in direct correspondence with the duration of the pulses as indicated in figure 15-1.

If the pulses have a duration of 1 ms we wish the servo shaft to indicate 0° and if the pulse duration is 1.5 ms, the shaft must position itself to match the corresponding input pulse duration. This relationship of pulse duration to shaft angular position can be represented by

$$\theta = (P_D - 1)\ 100$$

where

θ = Shaft Position (in degrees)

P_D = Pulse Duration (in milliseconds) (15.1)

For instance suppose we have a pulse duration of 1.4 ms. Then

$$\theta = (P_D - 1)\ 100$$

$$\theta = (1.4 - 1)\ 100$$

$$\theta = 0.4\ (100)$$

$$\theta = 40^\circ$$

Our shaft position would then be 40° for an input pulse duration of 1.4 ms. From this equation we can see that a short pulse produces a lower angular position than a longer pulse.

The complete block diagram of the servo circuit that we are about to look at appears in figure 15-2.

In this circuit a pulse is applied to the input and then takes two paths. One path

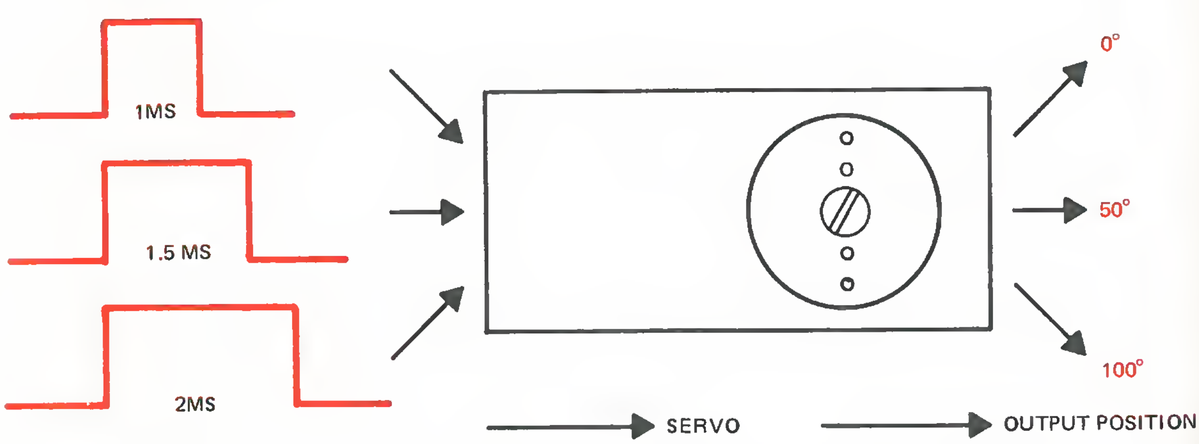


Fig. 15-1 Pulse Duration-Output Shaft Relationship

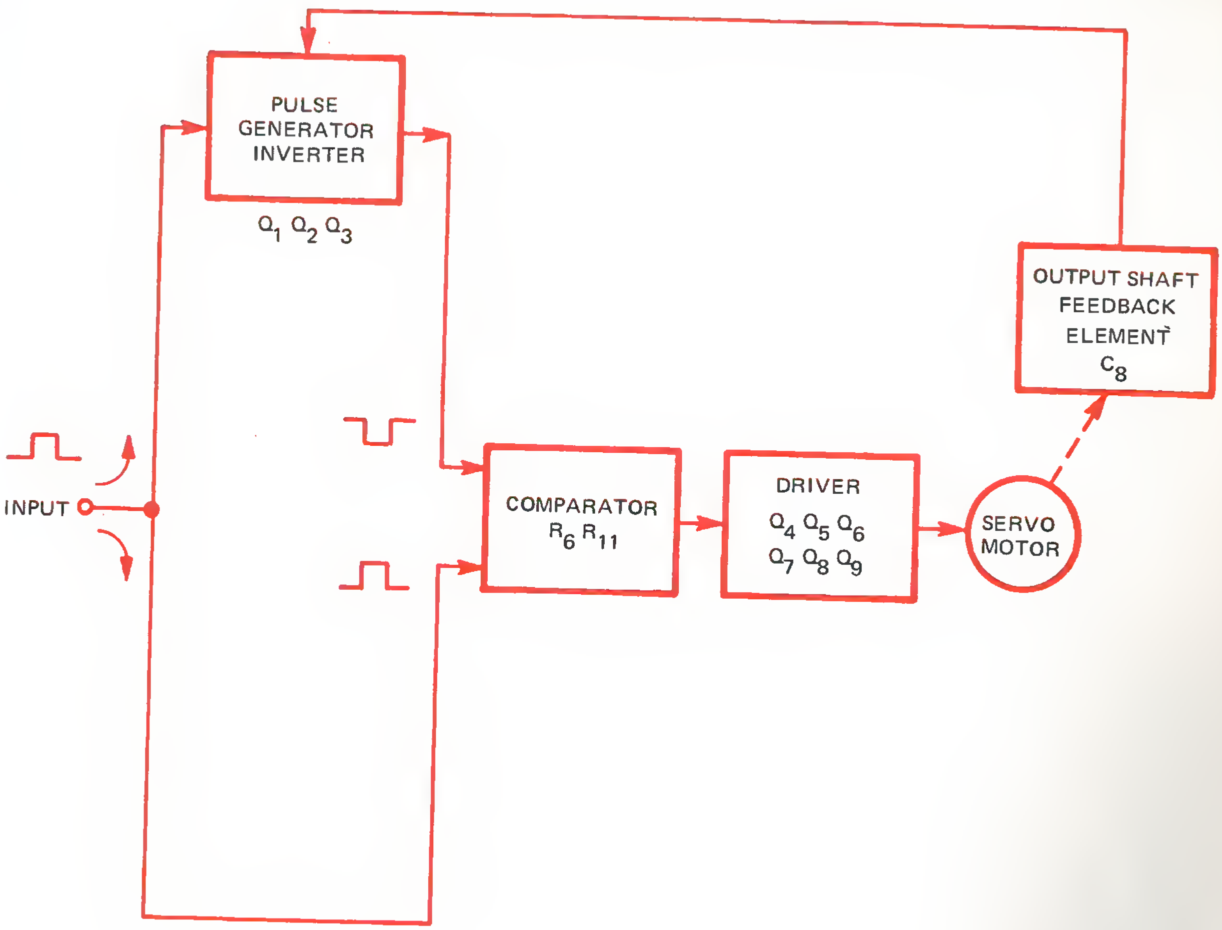


Fig. 15-2 Servo Block Diagram

triggers a variable duration pulse generator that produces a pulse whose duration is determined by the position of the output shaft. This pulse duration to shaft position relationship is the same as the relationship shown in equation 15.1. Solving equation 15.1 for P_D gives us the pulse duration from a given shaft position.

$$P_D = \frac{\theta + 100}{100} \qquad (15.2)$$

For instance suppose we have a shaft position of 40°. The pulse duration would be

$$P_D = \frac{\theta + 100}{100} \qquad P_D = \frac{40 + 100}{100}$$

$$P_D = \frac{140}{100} \qquad P_D = 1.4 \text{ ms}$$

Our pulse duration would then be 1.4 ms for an output shaft position of 40°.

This pulse generator produces a negative going pulse while the input is a positive going pulse.

The other path for the input pulse leads directly to a block called a comparator. The output pulse from our variable duration pulse generator is also applied to this block. The comparator receives these two pulses, compares their pulse durations and then produces a pulse with a polarity and duration equal to the difference between them. This is shown pictorally in figure 15-3.

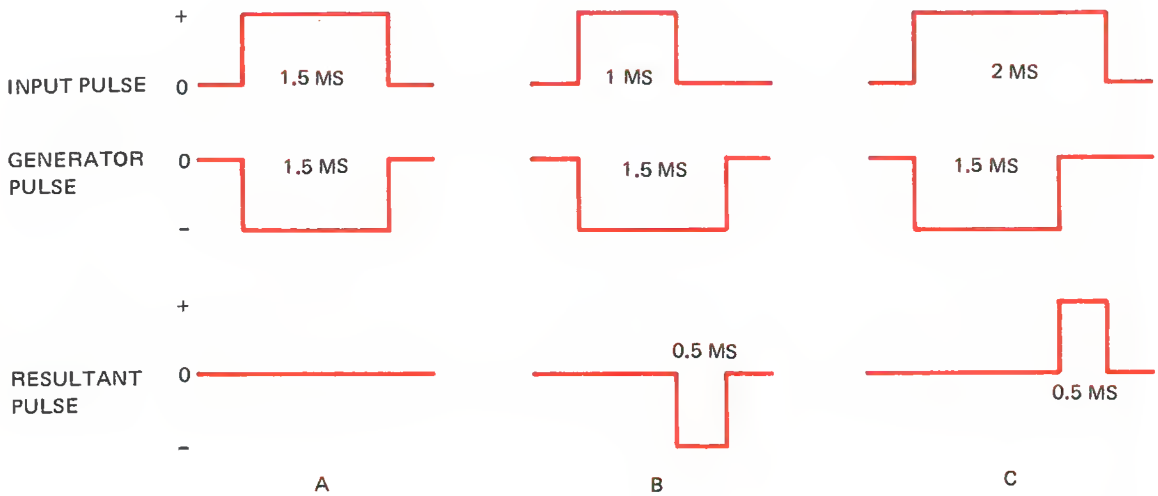


Fig. 15-3 Comparator Operation

Column A represents the comparator output. When both pulses have the same duration, this results in no pulse output from the comparator. In column B the generator pulse has the longest duration and is negative going as well. This results in a negative going pulse from the comparator. In column C the longer pulse is the positive going input pulse which results in a positive going output from the comparator.

In all the above cases the servo shaft was positioned at 50° and the pulse from the comparator indicated which direction and amount that the servo shaft must rotate in order to match the input. A negative comparator pulse indicates that the servo shaft should reduce its angular position. The duration of the pulse is representative of the angular error between the position indicated by the servo shaft and the position commanded by the input pulse. Equation 15.1 can be used to determine the angular error from the duration of the comparator output pulse.

If we had a circuit that would drive the motor counter-clockwise with a positive input and clockwise with a negative input, we would be satisfied. A circuit is shown in figure 15-4 that operates in this manner.

Q_1, Q_2 and Q_3 form the variable duration pulse generator we need, actually it is a monostable multivibrator. The pulse duration is determined by the time constant of R_1 and the parallel combination of $C_{8a+b+c+d}$ and C_9. This total capacitance is approximately 300 pF. Using the equation for determining the pulse duration of a monostable we find these values give us a maximum pulse duration of 2.3 ms.

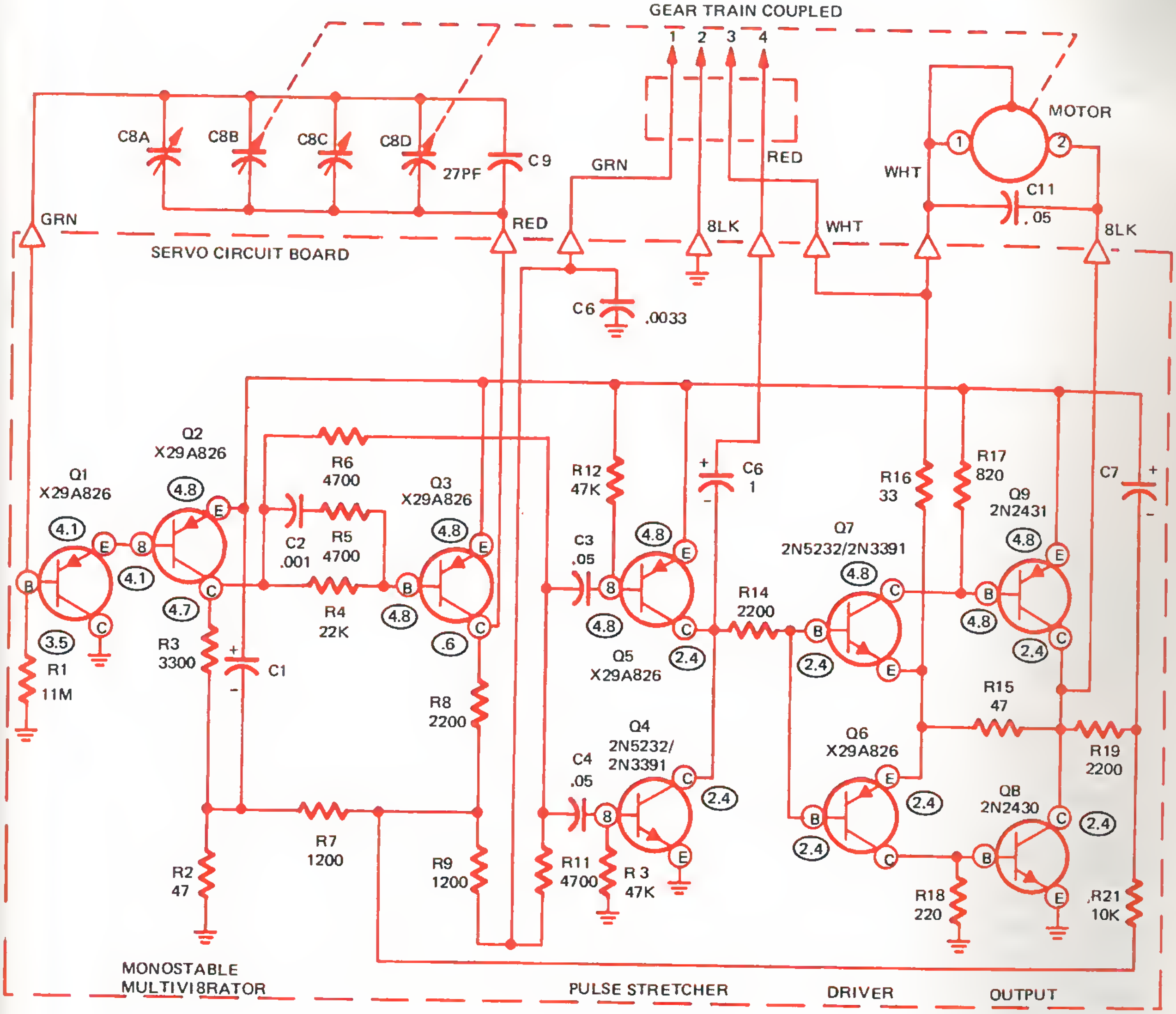
$$P_D = 0.69\ RC$$

$$P_D = 0.69\ (1.1 \times 10^7)(3 \times 10^{-10})$$

$$P_D = 2.3\ \text{ms}$$

A basic monostable is shown in figure 15-5. R biases Q_B normally ON and Q_A normally OFF. The input pulse turns Q_A ON and Q_B OFF and the RC time constant determines the time period which forms the negative going ouput pulse.

We can thus vary our pulse duration by controlling the value of either R or C or both.



(COURTESY HEATH CO)

Fig. 15-4 Comparator Operation

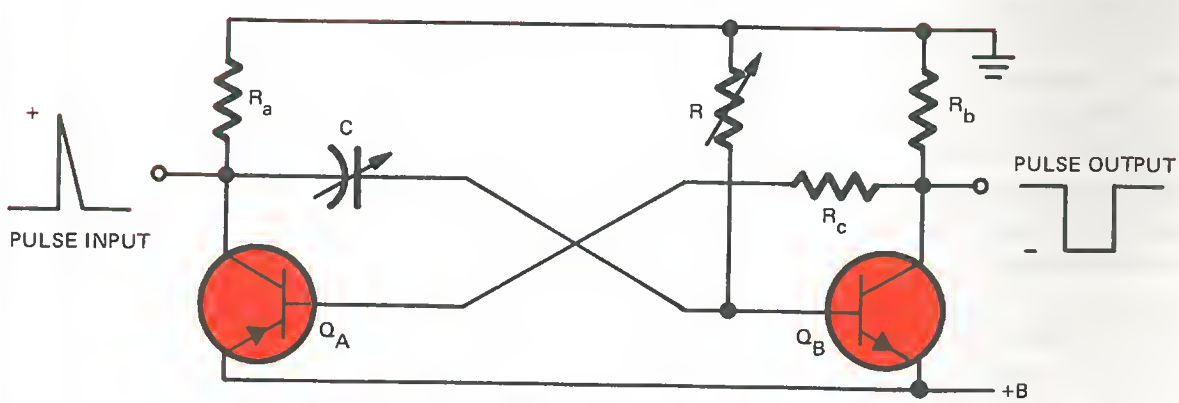


Fig. 15-5 Basic Monostable Multivibrator

In the circuit of figure 15-4, C_8 is varied and represents our feedback element of figure 15-2 and C of figure 15-5. The relationship between figure 15-4 and figure 15-5 is shown in the following table:

figure 15-4	figure 15-5
Q_3	Q_a
Q_1+Q_2	Q_b
R_1	R
C_8+C_9	C
R_8+R_7	R_a
R_3	R_b
$R_4+R_5+C_2$	R_c

The Q_1 and Q_2 circuit is designed to present a relatively high input impedance so as to have less effect on the R_1 $(C_8 + C_9)$ time constant. C_2, R_5 and R_4 form a base overdrive circuit providing faster mode change for the monostable.

The output of the monostable is fed through R_6 and the input pulse is fed through R_{11}. The two pulses are of equal amplitude but of opposite polarity, so since $R_6 = R_{11}$ the currents are also equal but in opposite directions. If the pulses have equal durations they will cancel each other and there will be no pulse transferred to Q_4 and Q_5 by means of C_4 and C_3. If the durations differ, either a positive or a negative pulse will occur and turn Q_4 ON if the pulse is positive or Q_5 ON if the pulse is negative. This action was described in figure 15-3 and takes place at the junction of R_6 and R_{11}. This type of junction is commonly called a summing point. C_4 and C_3 serve to transmit the resulting pulse to Q_4 and Q_5.

If the two pulses entering the summing point were not of equal amplitude they would develop an output pulse as shown in figure 15-6.

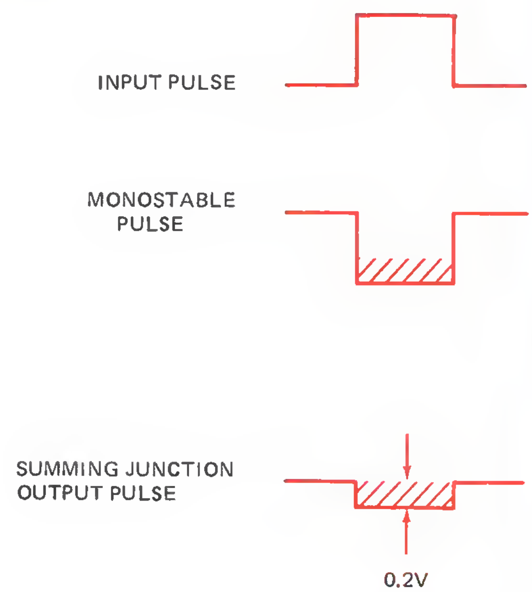


Fig. 15-6 Unbalanced Summing Point

The unbalance of inputs causes the shaded area of the monostable pulse to cause the shaded area of the summing point pulse to be developed.

R_{12} and R_{13} bias the bases of Q_5 and Q_4 respectively. Being silicon transistors they require approximately 0.5V to turn on. These two transistors are of PNP and NPN types and form a complementary amplifying stage feeding C_5. A pulse at the base of Q_5 must have an amplitude *in excess of -0.5V* for Q_5 to turn ON and discharge C_5. Likewise Q_4 must receive a pulse *in excess of +0.5V* for it to turn ON and charge C_5. This level of ±0.5V is called the threshold level and allows Q_4 and Q_5 to ignore any small unbalance in the comparator.

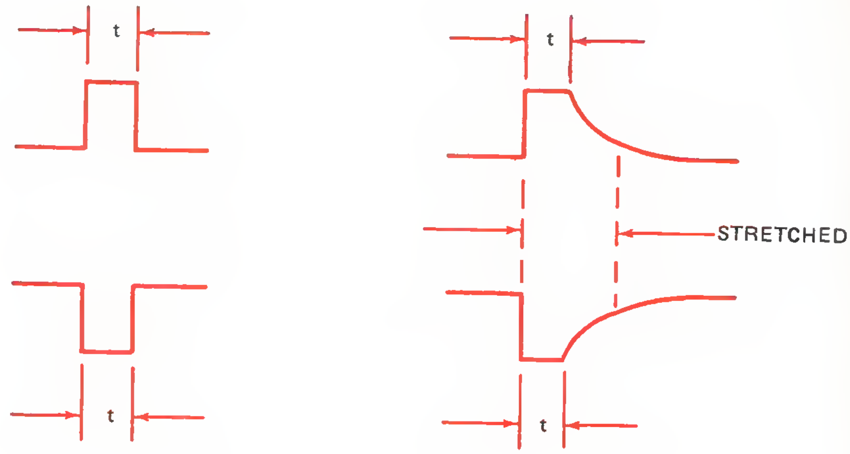


Fig. 15-7 Pulse Stretcher Waveform

C_5 stores the pulse passed by either Q_4 or Q_5. Q_6 and Q_7 form another complementary pair and conduct one at a time depending upon the level of the charge on C_5 in relation to the voltage level of their emitters which are tied to + 2.4V through R_{16}. Note that their bases are tied directly together which effectively biases them fully OFF. Their base current is supplied through R_{14} forming an RC time constant. Thus C_5 discharges at an exponential rate with a time constant of τ = RC = $(2.2 \times 10^3)(10^{-6}) = 2.2 \times 10^{-3}$ sec which in effect stretches the length of the pulse applied to the bases of Q_6 and Q_7 as shown in figure 15-7.

The period *t* is determined by the pulse duration and the exponential discharge curve by R_{14} and C_5. C_5 charges abruptly and discharges exponentially. Thus even a short pulse is stretched to approximately 2 ms to turn on Q_6 or Q_7. This stretched pulse is then further amplified by Q_8 and Q_9 to finally drive the servo motor. R_{15} limits the base drive current to Q_8 and Q_9 as it is passed by Q_6 or Q_7. C_{11} acts as a low impedance to noise generated by the brushes and switching transients of the motor. R_{19}, R_{21} and C_7 form a feedback network to the monostable Q_1, Q_2 and Q_3 to reduce overshoot and slow down the motor as it approaches the desired position. This prevents hunting or slow speed oscillation of the output shaft.

Note that Q_5, Q_7 and Q_9 are alternate PNP, NPN and PNP devices and form a complementary pulse amplifier. They are all OFF unless the servo is changing position and thus current drain is reduced at idle. Q_4, Q_6 and Q_8 operate in a similar manner. Q_8 applies negative signals to the motor while Q_9 applies positive signals. Thus the direction of rotation can be changed and is such that the feedback capacitor C_8 is driven in the direction that will cause the pulses at the comparator to cancel.

The receiver and 5 servos draw a maximum of about 1753 mA when *all* of the servos are moving against their normal load and about 16 mA when at their selected positions. The servo can change position in about 0.7 sec from one extreme to the other and this speed coupled with a low drain at idle results in a running time of approximately four hours under typical use with a fully charged battery pack.

MATERIALS

1 Radio control transmitter (Heath Co. model GDA 19-1 or equivalent)
1 Radio control receiver (Heath Co. model GDA 19-2 or equivalent)
1 Battery pack and harness (Heath Co. model GDA 19-3 or equivalent)
*1 Servo (on breadboard) (Heath Co. model GDA 19-4 or equivalent modified)
1 Oscilloscope
1 5kΩ potentiometer
2 2.2kΩ, 1/2w resistors

PROCEDURE

1. Remove resistors R_6 and R_{11} from the circuit board of the servo and replace them with the circuit shown in figure 15-8b. NOTE: *Do Not Overheat* Q_2 on C_3 and C_4. *Keep the leads to the pot short so as to reduce NOISE pickup.* This change will allow you to adjust the balance of the comparator.

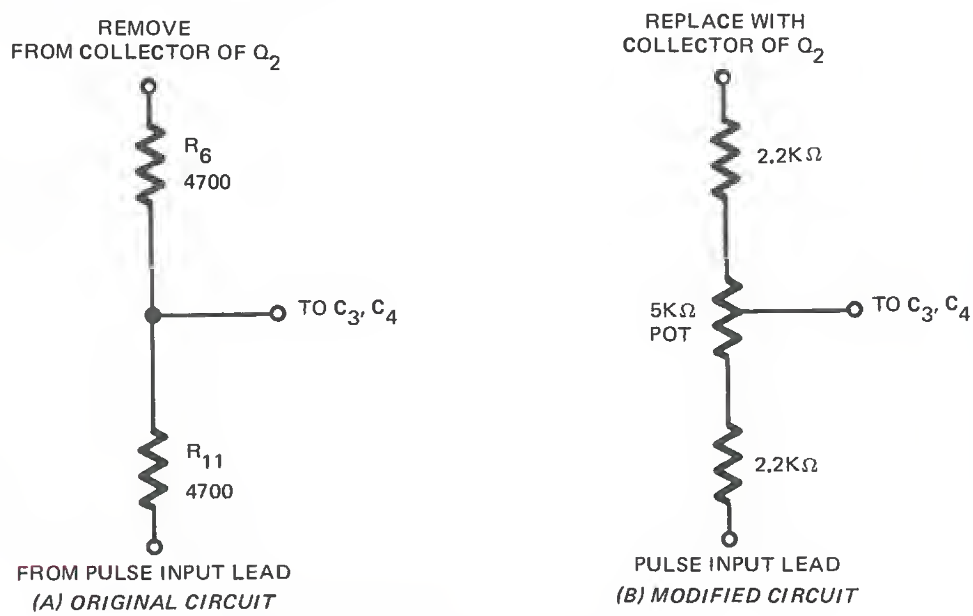


Fig. 15-8 Circuit Changes

2. Set the pot to the center of its resistance range as nearly as possible.

3. During the following disassembly of the servo gear train, *use extreme caution*. The mechanism is delicate. Do not over tighten the screws. Refer to figure 15-9 as a guide.

4. Remove the screw holding the output disk and gently remove this disk.

5. Remove the screws securing the gear case cover and gently remove it. Make sure the output shaft and gear do not come off with the cover. See figure 15-9 for guidance.

*This servo should be constructed on a 3 in. X 6 in. circuit board so that access to the test points is easy.

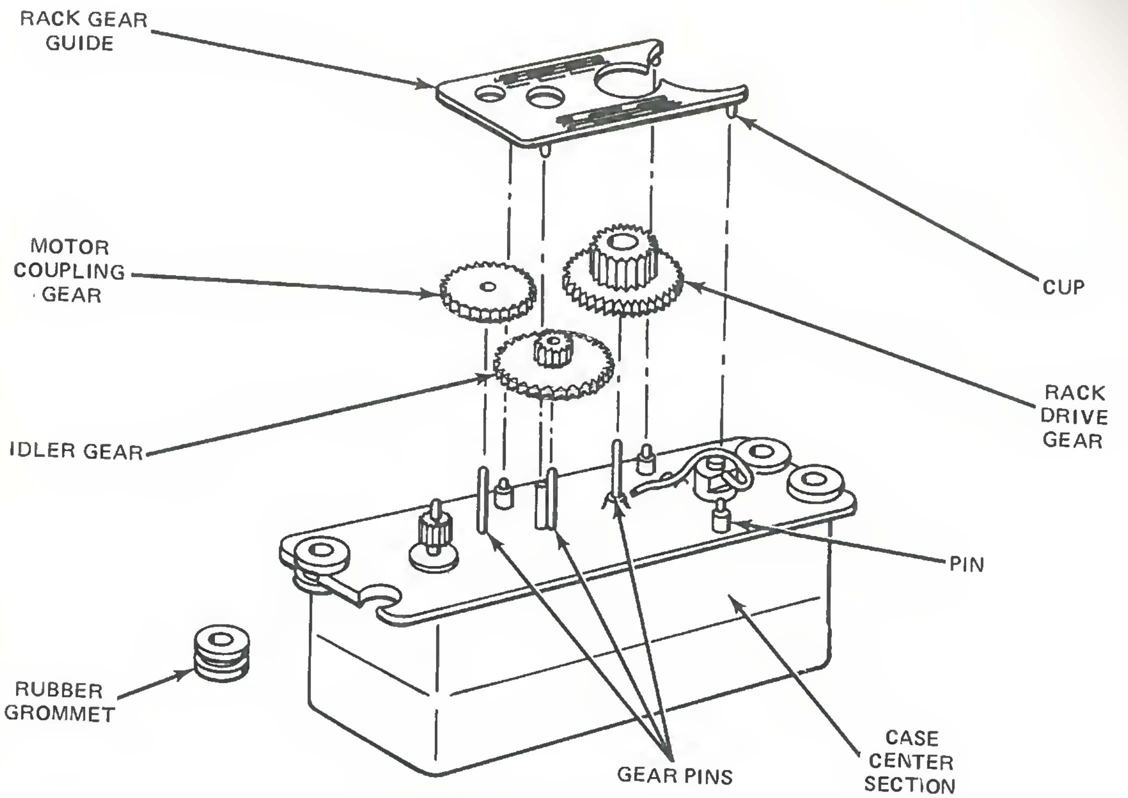


Fig. 15-9 Servo Gear Train

6. Remove the rack gear guide and the *motor coupling gear only*.

7. Replace the rack gear guide, the gear case cover and the output shaft disk. Replace the gear case cover screws and the output disk. *Tighten the screws only finger tight.*

8. You can now move the output shaft by hand. Set it to the center of its travel. DO NOT ATTEMPT TO FORCE THE OUTPUT SHAFT ON A COMPLETELY ASSEMBLED SERVO. They will become damaged as the gear train has a large reduction ratio and is not designed to operate backwards.

9. Connect the servo to channel 4 of the receiver. This is the channel operated by the vertical mode of the left hand control on the transmitter. This control has the centering spring removed and will thus stay in any position.

10. Check to see that the transmitter and receiver switches are *OFF* and then connect the receiver to the battery pack.

11. Make sure the transmitter antenna is fully retracted and then turn ON the transmitter. Observe the meter to verify operation.

12. Turn ON the receiver and move the channel 4 control to verify correct servo hookup. If the servo is operating and connected to the correct channel the motor can be heard to stop as the control is slowly moved past its center position. Observe the motor rotation through the hole in the case on the same side as the output shaft.

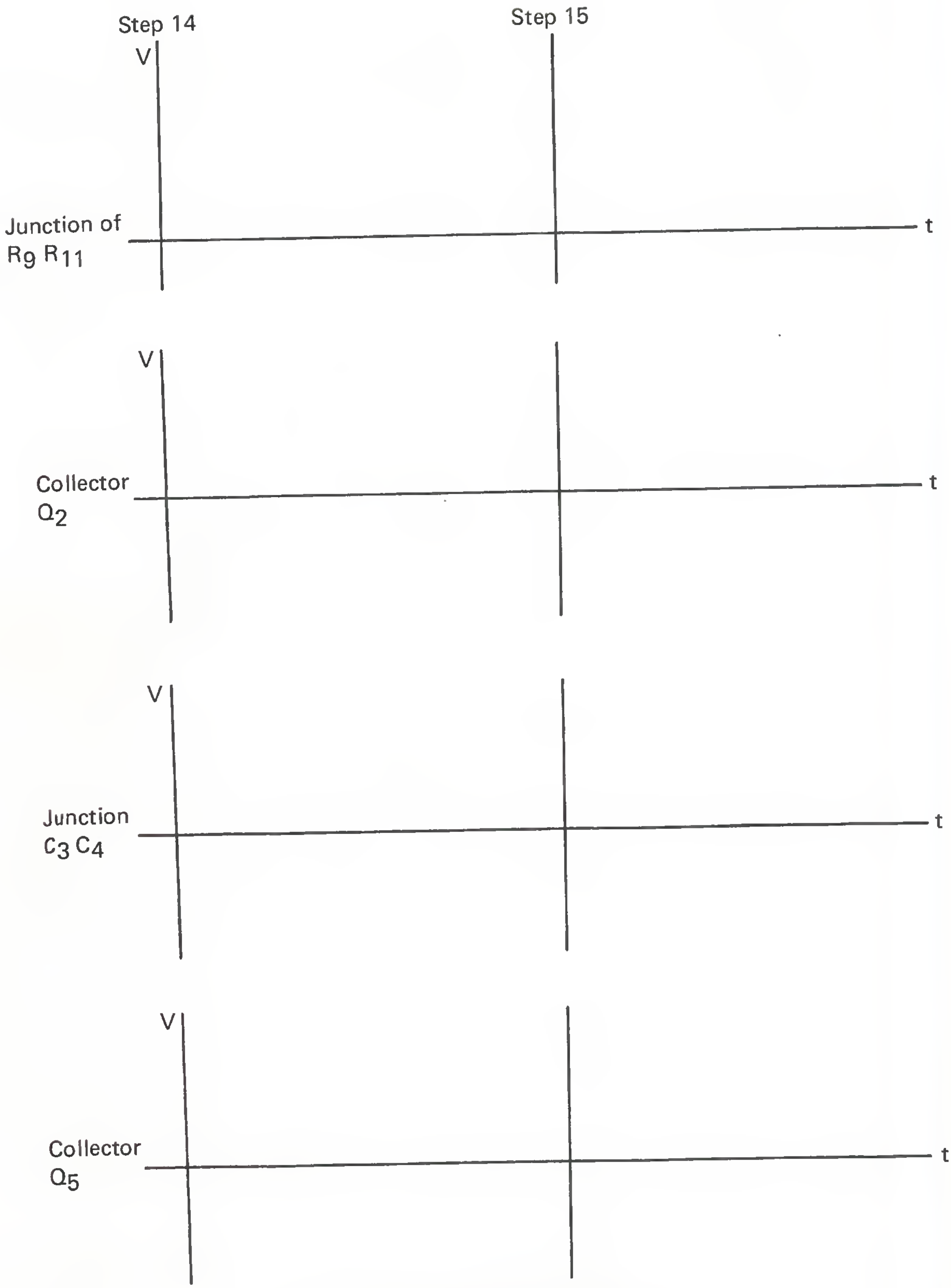


Fig. 15-10 The Data Tables

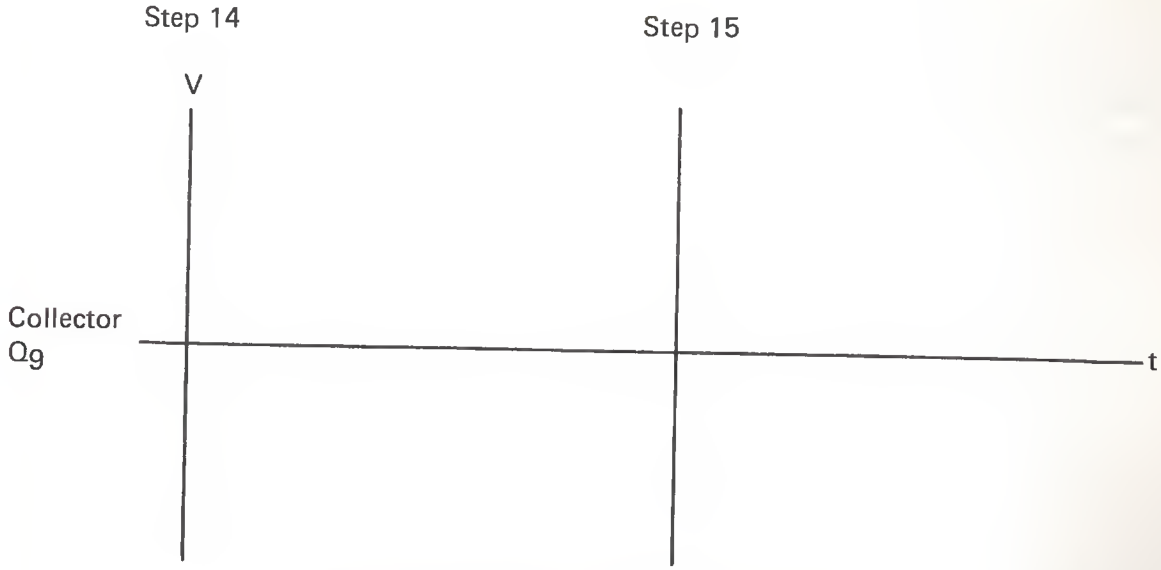


Fig. 15-10 The Data Tables (cont'd.)

13. Due to the circuit being spread out over a breadboard, the servo is apt to jitter and hunt from noise pickup. Adjust the control stick to stop the motor as completely as possible.

14. Observe and record the following waveforms. Sketch two complete cycles in the left half of the sets of axes shown in figure 15-10. Disregard random noise pulses and try to get the waveforms for the servo at the rest position.

15. Move the control to its furthest upward position and record the resulting waveforms in the right half of the axes used in step 14.

16. Move the control to its furthest downward position and observe the resulting waveforms.

17. Center the control as in step 13. Vary the pot you added to vary the balance of the comparator. Monitor the junction of C_3 and C_4. Note and record the maximum unbalance the servo can tolerate before losing control.

18. Adjust the comparator balance pot for best balance as observed on the oscilloscope. Note figure 15-6.

19. Turn OFF the receiver and then the transmitter.

20. Follow steps 3, 4 and 5 and replace the motor coupling gear.

21. Repeat step 7. Do not overtighten the screws.

22. Turn ON the transmitter and then the receiver and check to see that the servo functions correctly.

23. Operate the control and observe the reaction of the servo output shaft as it attempts to stop. Describe the action.

24. Turn OFF the receiver and then the transmitter.

ANALYSIS GUIDE. Discuss the merits of the complementary pulse amplifiers and their effect on current drain. What could be the advantages or disadvantages of a feedback capacitor versus a feedback resistor for determining the period of the monostable? Comment on the effect of R_{19}, R_{21} and C_7.

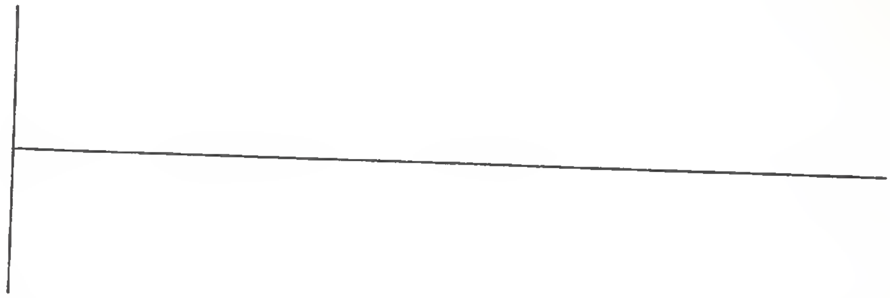
PROBLEMS

1. What is the function of C_9 in figure 15-4?
2. How does C_2 R_5 improve the operation of the circuit in figure 15-4?
3. From equation 15.2 determine the duration of the pulse for a shaft position of 73°.
4. What is the purpose of R_{17} and R_{18}.
5. What would occur if the leads to terminals 1 and 2 of the motor were reversed? Could the servo function, why or why not?
6. How would a transistor with a high I_{CBO} affect the duration of the monostable?
7. If the output shaft is positioned at 50° and a 1 ms pulse is applied to the input terminal, what will be the polarity of motor terminal 1 with respect to motor terminal 2?
8. Q_5 suffers a collector to emitter short. What will be the voltage at terminal 1 of the motor with respect to ground?
9. How would the servo act if Q_1 or Q_2 were to open up? Be specific.

EXPERIMENT 1 ____________ Name ____________

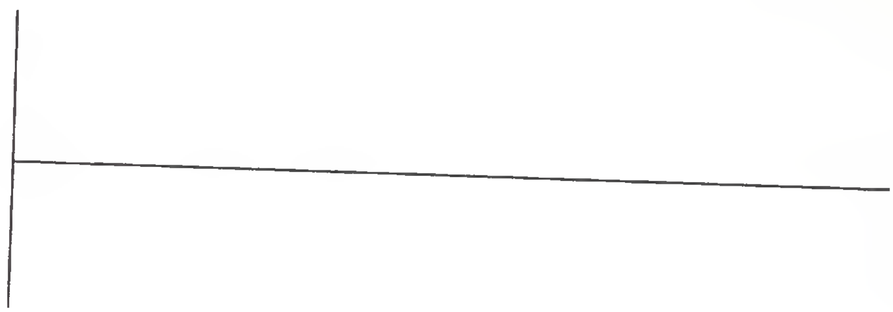
Date: ____________ Class ________ Instructor ________

Type of Receiver Circuit: ____________________



Generator Output Waveform

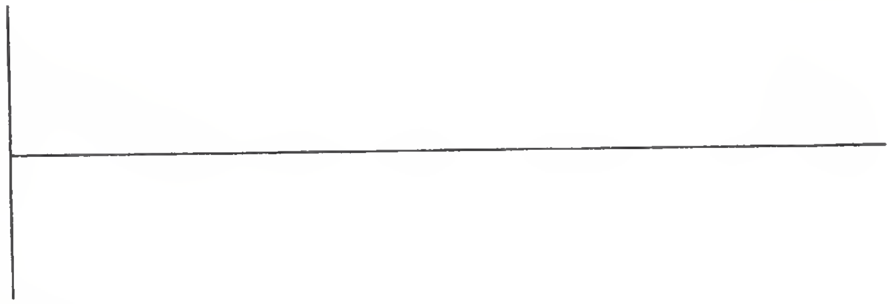
$f_{RF} \approx$ ____________________



Oscillator Waveform

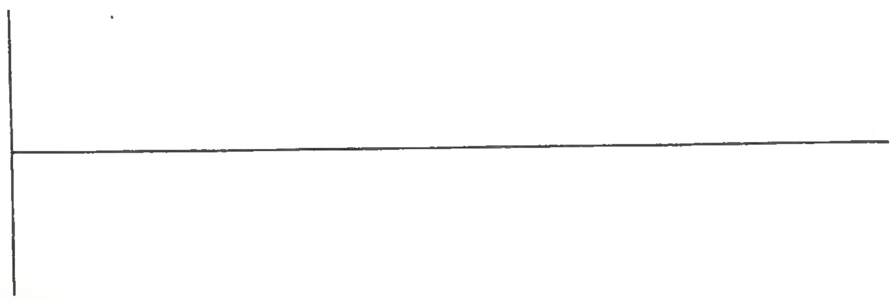
$f_0 \approx$ ____________________

Fig. 1-10 The Results



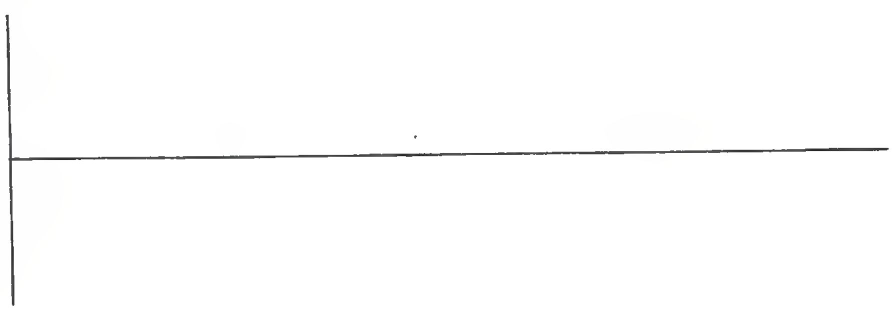
IF Amplifier Waveform

$f_{IF} \approx$ ________________



Detector Output Waveform

$f_m \approx$



Audio Output Waveform

$f_m \approx$

	Maximum Gen Output	Minimum Gen Output
1st IF		
2nd IF		

Base Voltages

Fig. 1-10 The Results (Cont'd)

EXPERIMENT 1 ______________ Name ______________________

Date: ______________________ Class __________ Instructor ______________

Fig. 1-11 Receiver Block Diagram

EXPERIMENT 2 ______________ Name ______________________________

Date: ______________________ Class ____________ Instructor ______________

Type and Value of

Compensation Components ____

____

FREQ	FIRST VOLTAGE	SECOND VOLTAGE	FREQ	FIRST VOLTAGE	SECOND VOLTAGE
50 Hz			6,000 Hz		
100 Hz			8,000 Hz		
200 Hz			10,000 Hz		
500 Hz			12,000 Hz		
1000 Hz			14,000 Hz		
2000 Hz			16,000 Hz		
4000 Hz			18,000 Hz		

Fig. 2-6 The Data Table

EXPERIMENT 3

Name ______________________

Date: ______________ Class ________ Instructor ____________

E_{max} = ________

E_{min} = ________

IF Waveform

E_{IF}	E_m	E_{dc}	R_{dc}	I_{dc}	A_2	R_m	I_0	I'_0

E_0 = ________

Modulation Signal

Fig. 3-6 The Data Tables

EXPERIMENT 4

Name ______________________

Date: ______________ Class __________ Instructor ______________

EXPERIMENT 5 Name ______________________

Date: ______________ Class __________ Instructor ______________

First Circuit

Base Voltage	Peak-Peak Output Voltage

Second Circuit

Base Voltage	Peak-Peak Output Voltage

Fig. 5-7 The Data Tables

EXPERIMENT 6 Name ____________

Date: ____________ Class ____________ Instructor ____________

f_{RF}	f_0	f_d	f_{IF}

f'_{RF}	f'_0	f''_{RF}

Fig. 6-7 The Data Tables

EXPERIMENT 7

Name ______________________

Date: ______________ Class ________ Instructor ______________

Type of Mixer Circuit = ______________

f_{RF}	f_0	f_{IF}

f'_{RF}	f_B	Δf

f'_{IF} = ______________

f''_{IF} = ______________

Fig. 7-8 The Data Tables

EXPERIMENT 8

Date: ____________ Name ____________

Class ____________ Instructor ____________

EXPERIMENT 9

Name ______________________________

Date: ____________________ Class __________ Instructor ______________

DATA WITH MODULATED INPUT

INPUT DATA

E_s = ______________

f_s = ______________

A = ________

OUTPUT DATA

E_1 = ______________

f_1 = ______________

E'_s = ______________

f'_s = ______________

E_2 = ______________

f_2 = ______________

E''_s = ______________

f''_s = ______________

E_3 = ______________

f_3 = ______________

Fig. 9-12 The Data Tables

DATA WITH UNMODULATED INPUT

INPUT DATA	OUTPUT DATA
E_s = ________ f_s = ________	E_1 = ________ f_1 = ________
E'_s = ________ f'_s = ________	E_2 = ________ f_2 = ________
E''_s = ________ f''_s = ________	E_3 = ________ f_3 = ________

A_V = ________

Fig. 9-12 The Data Tables (cont'd)

EXPERIMENT 10 ____________ Name ____________________

Date: ____________________ Class ________ Instructor ____________

Waveform from step 5

Type of waveform = ____________________

Step 8

E_o = ____________

f_c = ____________

Step 10

E_{max} = ____________ f_c = ____________

E_{min} = ____________ f_m = ____________

E_o = ____________ f_{USB} = ____________

E_s = ____________ f_{LSB} = ____________

Step 12

E_{max} = ________ f_c = ________

E_{min} = ________ f_m = ________

E_o = ________ f_{USB} = ________

E_s = ________ f_{LSB} = ________

Step 13

E_{max} = ________ f_c = ________

E_{min} = ________ f_m = ________

E_o = ________ f_{USB} = ________

E_s = ________ f_{LSB} = ________

Fig. 10-13 The Data Tables

Step 14

Step 15

E_{max} = ________ f_c = ________

E_{min} = ________ f_m = ________

E_o = ________ f_{USB} = ________

E_s = ________ f_{LSB} = ________

The overmodulated waveform

Signal Generator Waveforms

1

f = ________ type = ________

2

f = ________ type = ________

3

f = ________ type = ________

4

f = ________ type = ________

5

f = ________ type = ________

6

f = ________ type = ________

Fig. 10-13 The Data Tables (cont'd.)

EXPERIMENT 11

Date: ____________________ Name ____________________

Class ____________ Instructor ____________________

Fig. 11-18 The Data Tables

Channel No.	Main Control Range	Trim Range
1		
2		
3		
4		
5		

Fig. 11-18 The Data Tables (cont'd.)

EXPERIMENT 12 Name ____________________

Date: ____________________ Class ____________ Instructor ____________

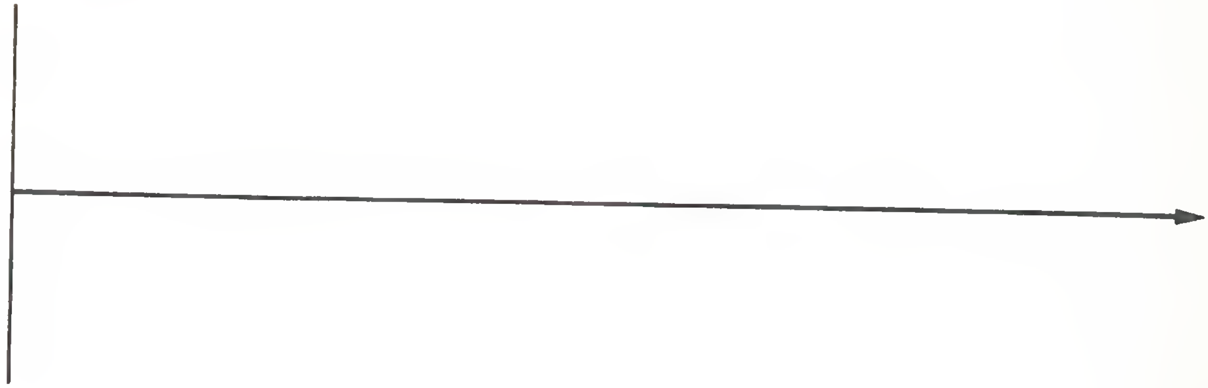


Fig. 12-14 Results from Step 11

	Channel Interval
R_{12} Normal	
R_{12} Max	
R_{12} Min	

Sync pause interval = __________

Fig. 12-15 Results from Steps 13 and 15

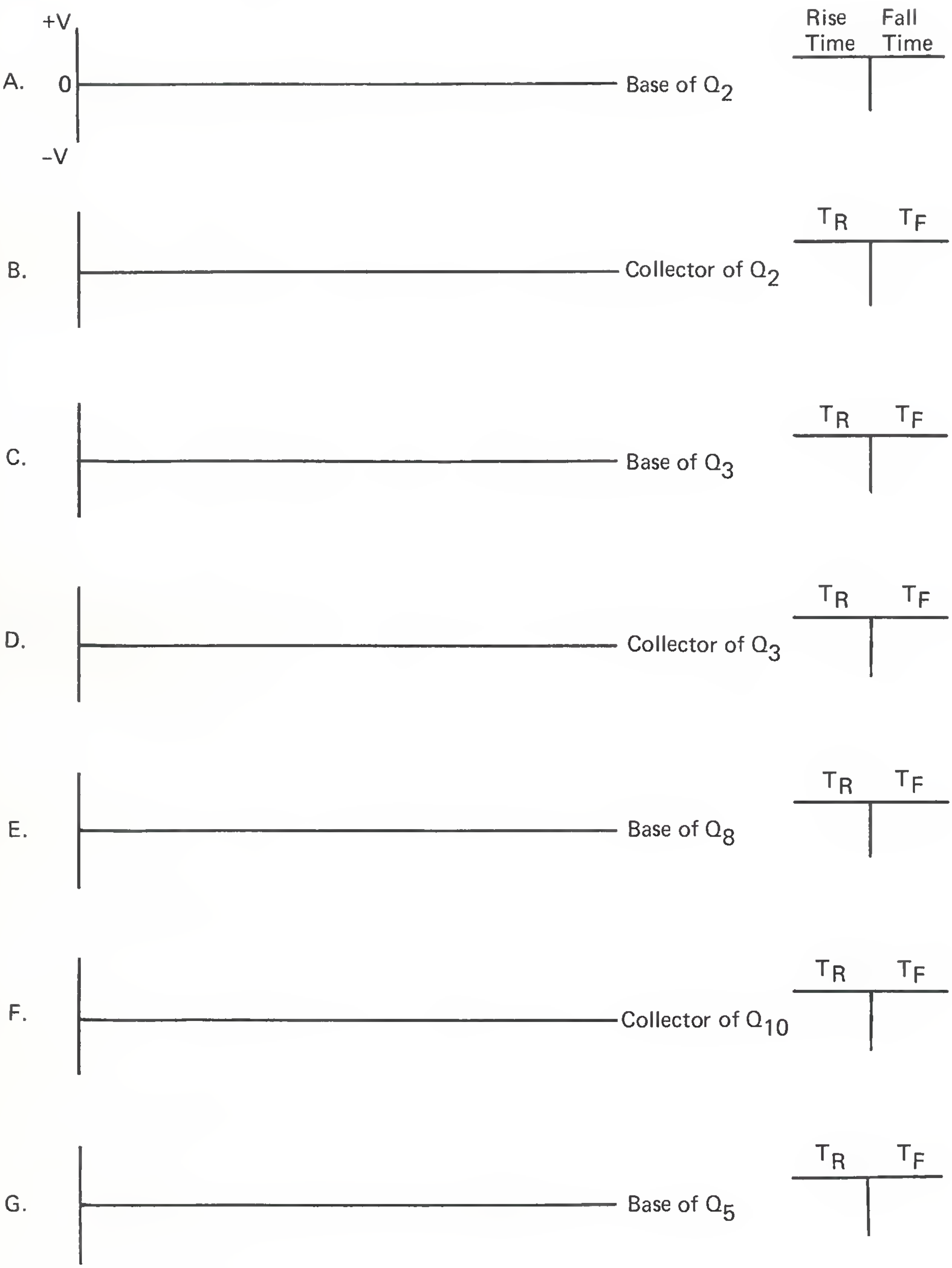


Fig. 12-16 The Data Tables

EXPERIMENT 13 Name ______________________

Date: ______________________ Class ____________ Instructor ______________

2. Compute the maximum current that the power supply will have to deliver to the circuit under worst case conditions. Show calculation and record.

I_{max} = ____________

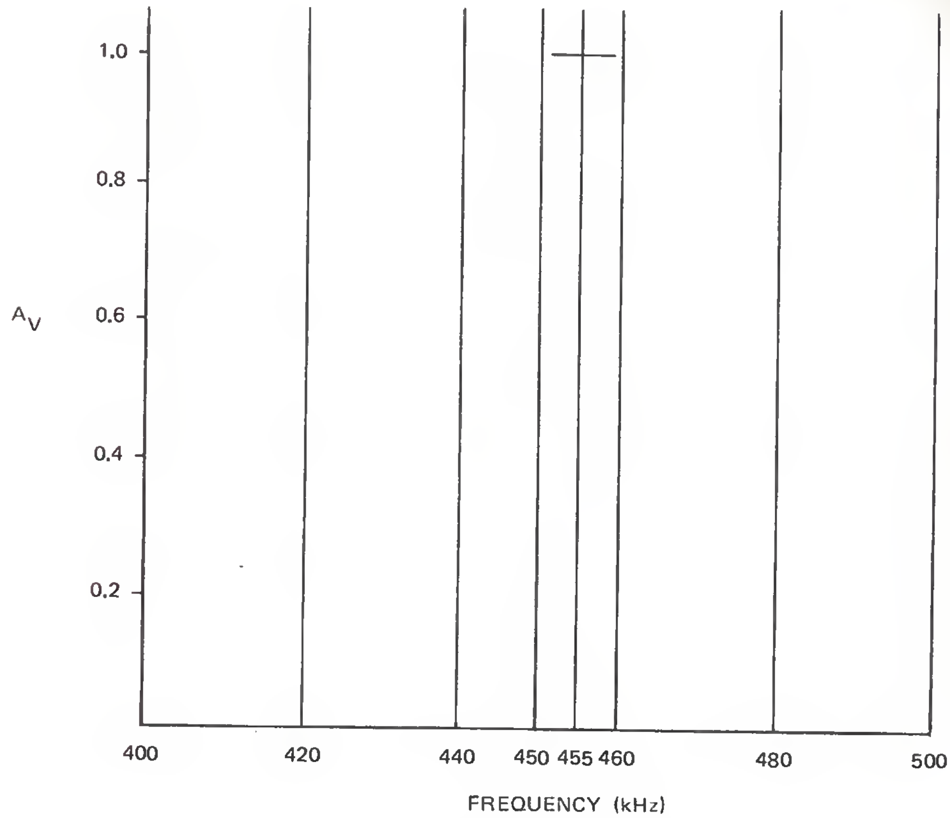


Fig. 13-10 Response Curve

12. From the data obtained determine the maximum voltage gain.

A_V = ____________

EXPERIMENT 14 Name ______________________

Date: ______________ Class __________ Instructor ______________

TEST POINT	WAVEFORM	RISETIME	FALLTIME	AMPLITUDE
Base of Q101				
Collector of Q102				
Collector of Q103				
Collector of Q104				
Cathode Gate of SCS-101				
Anode Gate of SCS-102				
Ch. #1 Output				

Fig. 14-17 The First Data Table

Collector Q102	
Ch. 1	
Ch. 2	
Ch. 3	
Ch. 4	
Ch. 5	

Fig. 14-18 The Second Data Table

EXPERIMENT 15 Name ______________________

Date: ______________ Class __________ Instructor ______________

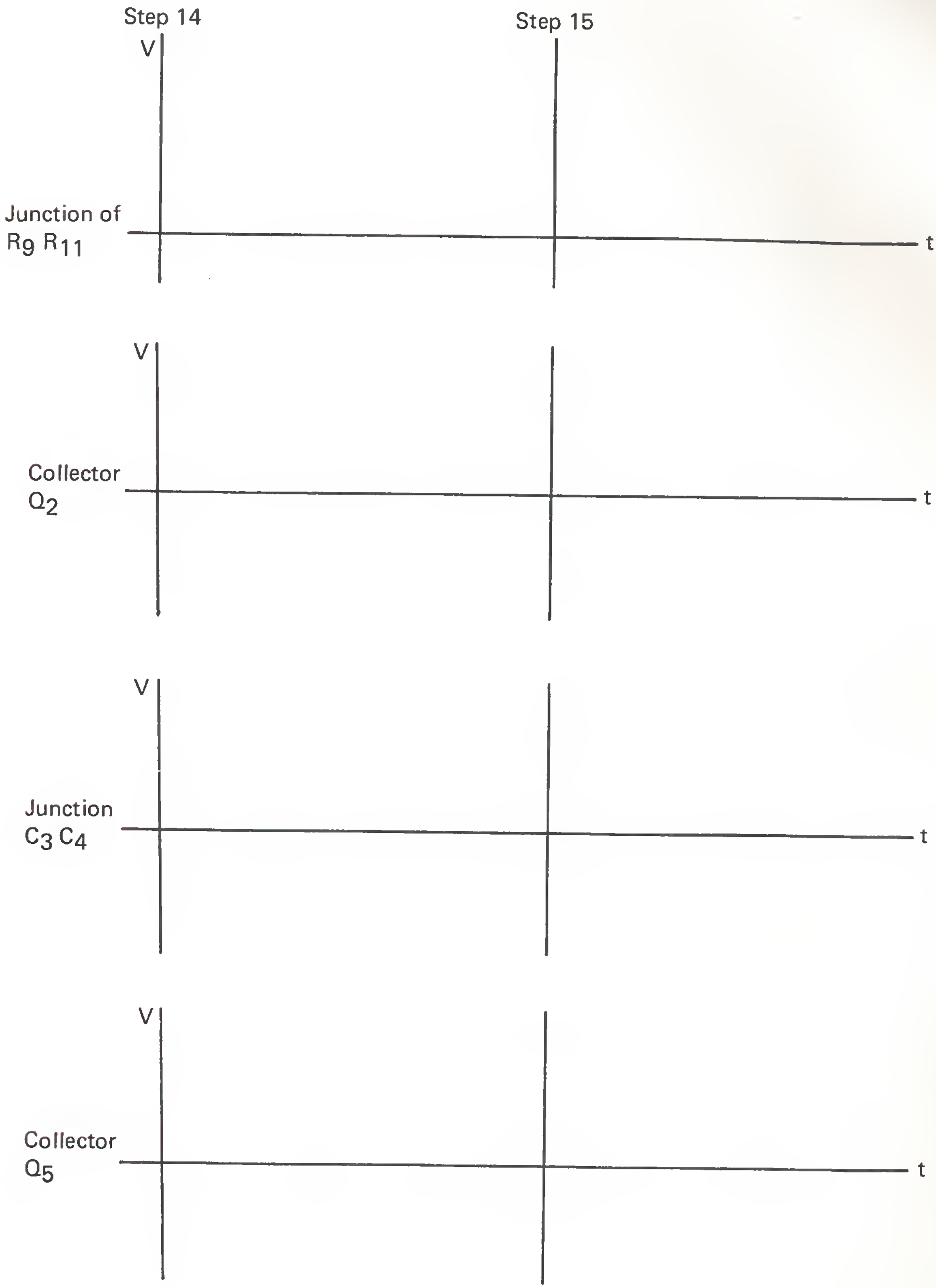


Fig. 15-10 The Data Tables

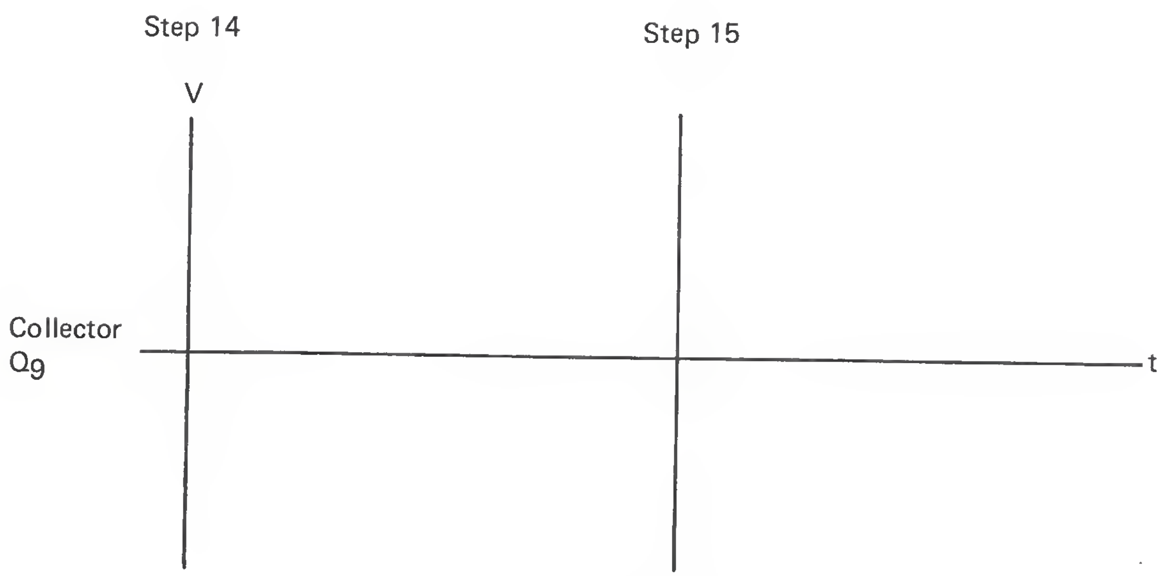


Fig. 15-10 The Data Tables (cont'd.)